The Business Telephone System Specialists

GDS

Hybrid IP Series

IP Extension Solutions

Technical Guide

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This guide was written and compiled by Russell Campbell at Auto Telecom Australia
For the Auto Telecom Group of companies and Hybrex Partners worldwide.

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Section 1

G2-ISU3 IP Extension Card for GDS

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G2 - ISU3 Description

The G2-ISU3 Card when installed, configured, and connected to an intranet, or the Internet, provides a Hybrex GDS system with the capability of hosting SIP IP telephones as system extensions. These SIP IP phones can be located remotely at any location with a broadband intranet or Internet connection and will function in much the same way as any other system extension. If generic SIP IP phones, Softphones, or even WiFi SIP phones, are used the functionality will be similar to an analogue extension phone connected to the GDS. If Hybrex IP handsets are used the added functionality of BLF, DSS and function keys, will be available in much the same way as a standard DK series system handset. These remote IP extensions are treated in the same way as system integral extensions in GDS programming. So for example they can be placed in ring groups, transfer calls, use the centralised Voicemail of the GDS, and access available trunks on the GDS etc in the same way as other extensions.

The G2-ISU3 card comprises a specific software version run on the Hybrex G2-VIUS hardware platform.
The ISU3 essentially acts in the same way as a SIP server in that remote IP phones "Register" to the ISU3, after which the phones are seen as "extensions" by the system and receive appropriate signaling.
Similarly to a G2-VIU the ISU3 installs in any trunk slot in a GDS but the ISU3 requires an 80 port cabinet base.

When installed and configured a G2-ISU3 provides:

- Capacity of 3 concurrent voice channels into the host GDS system.
- Capacity of 8 connected remote IP extension phones generally.
- Capacity of up to 24 connected extensions when the ISU3 is located in the last cabinet trunk slot (9) and no PRI ISDN card (G2-PIU) is installed in the same cabinet.
- Available Codecs for voice are G723(6.5), G729, G711u, G711a.
- Post Dial DTMF : currently only SIP INFO style is fully supported.

G2 - ISU3 Hardware

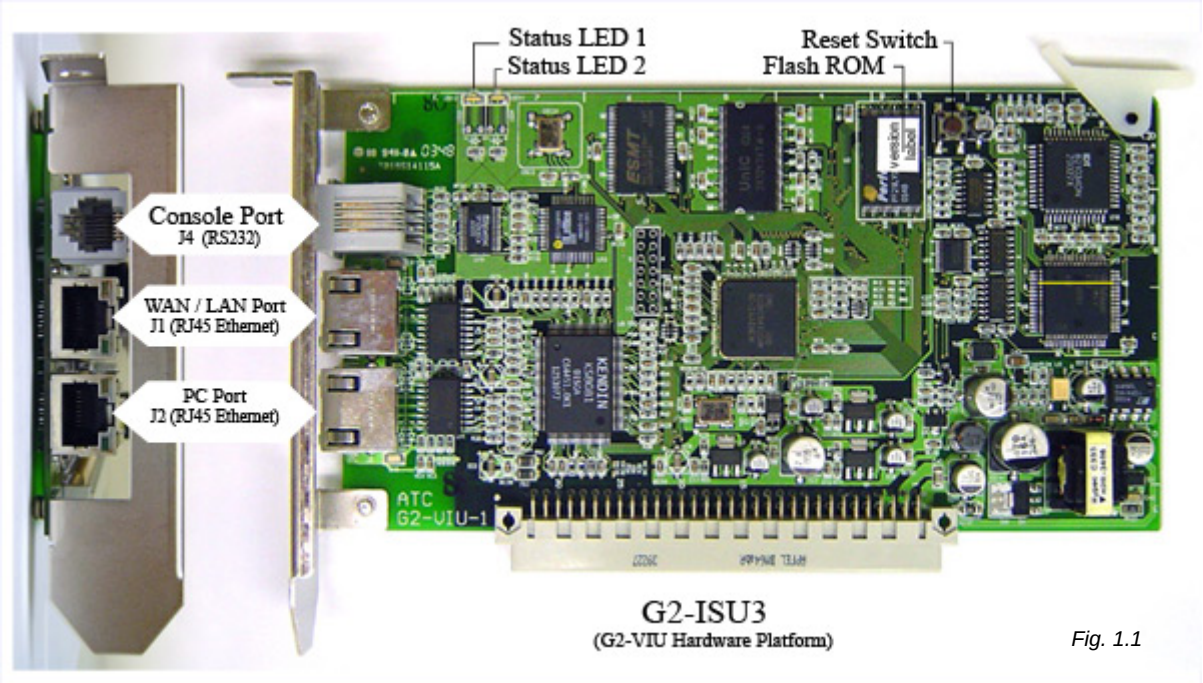

Fig. 1.1

Ground Rules

Before describing ISU3 setup and other items there are basic requirements that must be met if deployment and functionality is to be successful. Following that the various configuration options will be described as the choice of configuration will affect the ISU3 card settings made.

Network Requirements:

Using the Internet generally as the conduit for VoIP deployment has some level of uncertainty regarding the transport of voice traffic. Voice traffic is “real time” traffic reliant on minimal, preferably constant, delays and minimal loss of data “packets” for a quality result. What this means is that network congestion and other factors can markedly affect quality. In general data transport across the Internet is managed on a “best effort” basis so the expectation that VoIP telephony will be equivalent in quality to the circuit switched methods known to date, whilst achievable in many cases, is not to be expected as a given. Short of attaining specific Service Level Agreements from able Internet Service Providers (ISP), the first line of defense against the above is to choose where possible a reliable ISP who has preferably a low contention ratio (ratio of bandwidth sold out front to what their backbone connection is capable of). This includes the first point of connection: the DSLAM at the local exchange - if the ISP owns it the situation is better than otherwise. If the ISP is able to qualify traffic according to Type of Service (ToS or “Diffserv”) this is better because it allows voice traffic priority at the outset. Apart from the other considerations noted below in general the old maxim applies : “you get what you pay for”.

Dedicated Connections are recommended:

This applies to any form of VoIP where multiple channels are to be carried. For VoIP trunking, and for the subject of this document the ISU3 card, it is *strongly* recommended that the Internet connection for this equipment is dedicated to voice. Where an Internet conduit carries other data, for computers, email, etc. it is general experience that there will always at some point be data contention resultant in voice quality decline, breakup, and even decimation in the worst cases. Where the connection is asymmetric (ADSL) the most vulnerable direction is upload as this will have the lowest bandwidth. In locations where single IP extensions are deployed the dedication of the connection is impractical—in this case however the user is usually in a situation where there are few other computers etc connected and they are aware and to some extent in control of the traffic situation. However in a larger office where one or more ISU3’s are installed—dedicate the Internet connection for voice, your customer will thank you for it.

Bandwidth Requirements:

The bandwidth required for VoIP is dependant on the codec employed for the connection. The codec is essentially the compression algorithm used for the voice data. Whilst the G711u and G711a codecs are available, supported, and represent virtually no compression - giving the highest voice quality - they have a high bandwidth price: 90 Kbps per channel, and are relatively intolerant of packet loss: i.e. quality degrades quickly with packet loss. At the other end of the scale G723-6.5 is also available to use and requires on 22Kbps per channel. G723 however is only useful where there the network has minimal delays and losses, e.g. LAN only deployment, as it quickly degrades with increase of these factors. The recommended codec, providing the best performance/bandwidth trade-off is G729. G729 requires 30Kbps per channel. So one ISU3 card, with a 3 channel capacity, is going to require 90Kbps for voice plus some allowance for command and other network traffic. It is recommended then that the minimum dedicated ADSL account for one ISU3 card , using G729 codec, is a 512/128 account. From experience this type of account will usually provide something like ~480Kbps down and ~110Kbps up which is ample for one ISU3. If multiple ISU3’s are to be employed the bandwidth requirements will increment in proportion with the likelihood of a “symmetric” ADSL account (e.g. 512/512) being the preferred supply option.

ISU3 Network Configuration Options

There are four basic network configurations for the ISU3. Which of these is chosen will depend on where the IP extension devices are to be deployed and the network facilities available.

The four types are:

1. LAN or VPN Intranet only
2. NAT Configuration
3. PPPoE Configuration
4. Public IP Assignment

Network Type 1 : LAN or VPN Intranet Only

This type would be chosen where IP extension devices are to be deployed across a LAN, or at other locations via a Virtual Private Network (VPN), only. An example would be remote extensions distributed across an education campus, or a multi-site business where a VPN was available. A note of caution for both examples would be: since the network is likely to be shared with data traffic that specific attention needs to be paid to traffic prioritization, i.e. Quality of Service (QoS) issues - usually by combination of router/managed switch programming and ISU ToS settings - so that voice traffic has high, or highest, priority. Establishing connections across a LAN is basic in that the LAN router manages the traffic internal to the LAN – to and from the privately addressed devices. Traffic across a VPN is managed similarly in that a VPN is like a segmented LAN where the routers concerned will pass traffic to the relevant segment according to device addresses - the addresses again being private. This type of network will not be able to establish connections with any IP extension devices that are *not* located in/on the particular LAN or VPN. I.e.: IP extension devices will not function with this setup if connection is attempted from an arbitrary location on the Internet.
For this network type the prime requirement is the ISU3 must be assigned a fixed IP address.

Network Type 2 : NAT Configuration

This type of network will be the most commonly implemented type as it offers the best advantages for the resources deployed. IP extension devices can be located at any arbitrary location with a broadband Internet connection. These IP 'stations' can connect from behind most modem/routers (Network Address Translation, or NAT, firewalls) automatically, without any special settings having to be made at that location. The IP extension devices are also location portable - they can be moved to wherever desired and when plugged back into a broadband Internet feed, register with the ISU3 and are once again a live extension of the GDS.
The ISU3 device (or devices) are connected to the Internet via a standard modem/router. With a series of port forward settings made on the router, and the known WAN IP address of the Internet connection along with some other settings programmed into the ISU3, the ISU3 behaves as if it were directly connected to the Internet, hence allowing Internet distributed IP extensions. Multiple ISU3's can also be configured on the one account using variant port sets.
The requirements of this setup are :

(a). The Internet access account used must have a "static" (not dynamic) WAN IP address. This is because the ISU3 must be at a known, constant, and contactable, address for the IP extension devices to connect to. Note: in the absence of a static IP there are other measures that can be employed such as DynamicDNS but these require specific equipment and are not without possible problems - static is the recommended option. (See appendices for further information)

(b) The type of modem/router employed can have significant effect on performance - see appendices for recommended types.

There is another advantage for the NAT configuration in that the broadband connection, set up for the ISU3, can also be configured to allow remote programming/maintenance of the ISU3 card itself and even the GDS MPU if desired. In this way one of the hurdles of integrators - that of resistance of the IT departments of some organizations to allow any form of connection to their LAN of equipment they are unfamiliar with (e.g. GDS MPU) - is overcome. This is of course unless the use of GDS CTI (HybrexCAS, TAPI etc) forces that issue.

Network Type 3 : PPPoE Configuration.

Like NAT configuration, PPPoE configuration allows deployment of IP extension devices across the Internet to any location with a broadband Internet connection, with location portability. PPPoE stands for Point to Point Protocol Over Ethernet and is the common form for clients to connect with an ISP. In the PPPoE process the client (in this case the ISU3) is assigned the public address (real Internet IP) of the account. This is one way of obtaining a real IP which is a prime requirement for a “registrar” such as the ISU3, although this has been somewhat overtaken by the ISU3 programming now allowing IP sharing as the NAT configuration demonstrates.
In PPPoE mode the ISU3 is connected to the Internet feed via a modem set in bridge mode. There is a mandatory requirement of a “static” IP for the Internet connection meaning the Internet feed will most likely be ADSL or variant, as this requirement is not often met by other forms of connection (e.g. Cable in particular). This static IP is then the address of the ISU3 for the purpose of IP extension registrations and connections.
Because the ISU3 is then the account client and does not contain a switch or router the Internet account is therefore automatically dedicated to this one ISU3 card. This rules out this method for multiple ISU3 installations (unless a separate account is used for each card which may only be viable if 512/128 accounts were all that was available).
Advantages are few but PPPoE mode does provide direct connection to the ISU3 for remote programming/maintenance and a clean and simple setup where one ISU3 is to be deployed.

Network Type 4 : Public IP Assignment

As per types 2 & 3 the assignment of a public IP (real Internet address) to an ISU3 allows deployment of IP extension devices across the Internet with aforementioned advantages. Public IP assignment requires the availability of real IP’s, usually obtained by leasing a group of them known as a “subnet”, which are then routed onto the network containing the ISU3/s. If this approach is taken the mandatory “static” IP requirements are automatically met, with a 4 IP subnet being the minimum requirement for one ISU3 (first and last addresses cannot be used, middle two: one is assigned to ISU, the other is usually assigned to the router as a gateway). Where more than one ISU3 is to be deployed the next available subnet size is 8 IP where there are 6 available addresses. This information is relative to situations where you, the integrator, are handling the entire setup and have the requisite IT experience to do so. There will be situations where this method is employed because the IT department/support for the client network, where the ISU3/s are to be situated, are able to supply public IP/s for use. If they are able to supply this service then the relevant details (IP address/s and gateway IP) will be provided by them.
The caveat is that if public IP’s are used this implies a larger network where voice will most likely be sharing bandwidth with data so Quality of Service (QoS) measures must be employed to maintain voice integrity. This is usually done by combination of router/managed switch programming and ISU3 ToS settings - so that voice traffic is allowed high, or highest, priority.

Installation of G2-ISU3 in a GDS

GDS Requirements:

MMD80 - 80 port GDS cabinet - is required as the base system.
Installations in converted 64 port cabinets or 40 port cabinets are not currently supported.

GDS Software: minimum software required is MPU: G3-E050v, IPU: W0bmbb at this time.
Please update the GDS to the above or later software versions before installation of G2-ISU3.

Preliminary GDS programming settings:
Please set :
05-12-01 = 1 - (hangup) - this enables the remote extension transfer method.
05-14-06 = 3 - this delays Caller ID, enabling proper reception by SIP endpoint devices.

ISU3 Placement :

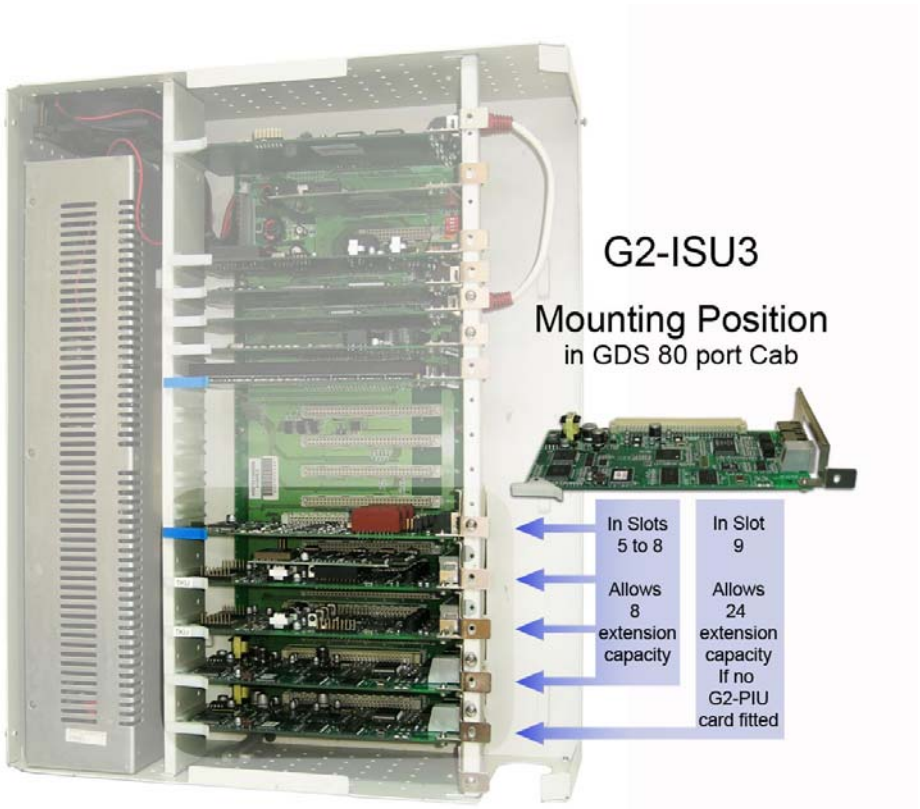

Fig. 1.2 ISU3 Mounting Positions

Power down the GDS and install the G2-ISU3 card in the selected slot.

Power the GDS back up and after allowing a couple of minutes for the system to initialise, connect and log onto GDS System Programming where you can check the ISU3 has been recognised by the system by selecting : Technical Program > Card Installation
For example:

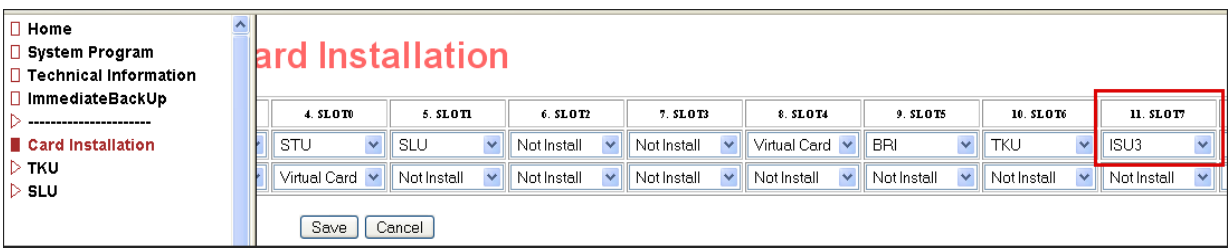

Fig. 1.3 GDS Technical Program - Card Installation

GDS Programming for ISU3

As previously indicated the ISU3 can be installed in any trunk slot in an 80 port cabinet. When the ISU3 is installed in any of Slots 5 through 8 : 8 extensions can be registered on that card. If the ISU3 is installed in Slot 9 : 24 extensions can be registered except when a G2-PIU card is installed in the same cabinet in which case available extension positions revert to 8.

Extension numbers are set up in System Programming > Mode 21 : Port/Station Number.

Select the slot the ISU is installed in and enter the desired extension (station) numbers for the remote IP extensions **(1).** Click [Save] when done.

Where an ISU is located in Slot 9, and no PRI (G2-PIU) card is fitted in the same cabinet, the extra station numbers available are then programmed into the following virtual slots (xa,xb).

As and when the IP extension devices (SIP phones) individually register successfully with the ISU3 card the "Type" field will show "IP Phone" for each registered device **(2).**

For Example :

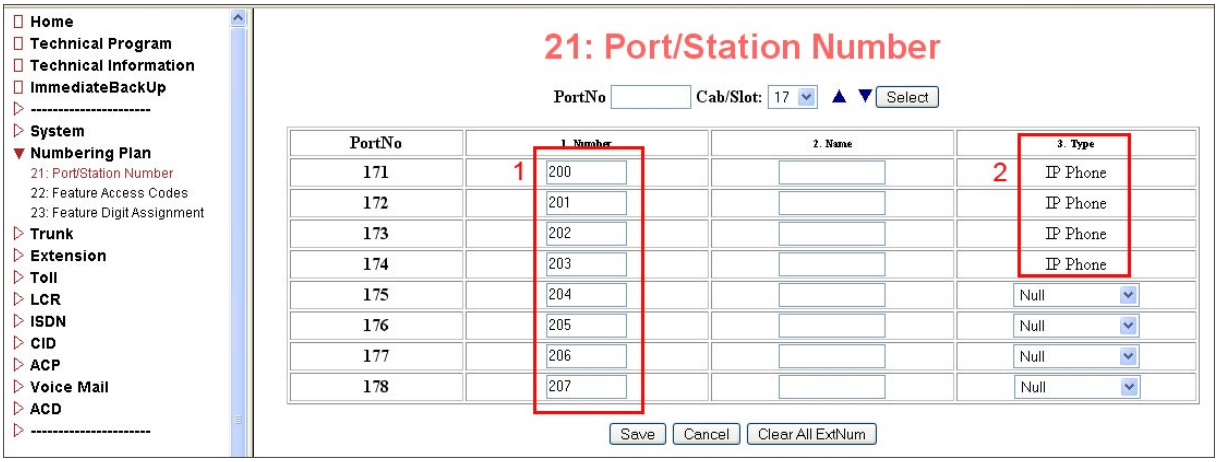

Fig. 1. 4 GDS programming - Mode 21

Having determined and set the extension numbers for the remote IP extensions in Mode 21 these extensions can then be treated the same as any other extension in GDS system programming. Bear in mind the (user) functionality of any SIP endpoint devices, other than Hybrex IP handsets, will be same as an analogue (SLT) extension.

G2-ISU3 : Connecting to, and Login to Programming

As with all VIU platform cards the ISU3 is setup by programming via its Web interface. This Web interface is available via a Web browser on any computer (PC) connected to the ISU3, either directly connected via a "crossover" Ethernet cable to the LAN port of the ISU3, or by standard Ethernet patch cables via a router or Ethernet switch to the ISU3 LAN port.

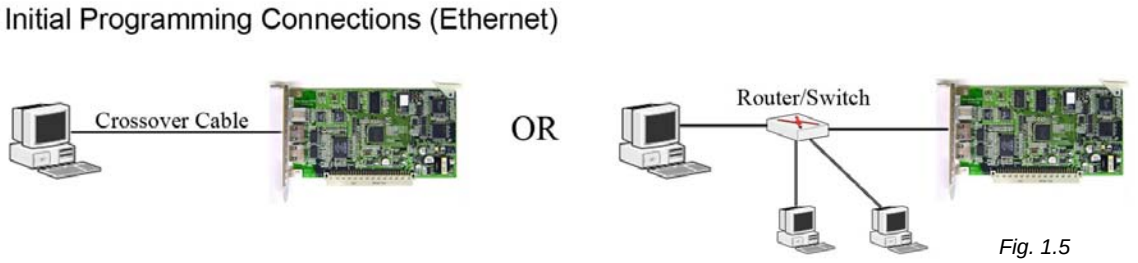

Fig. 1.5

The only requirement apart from the above cable type is that the Ethernet (IP) address of the PC in use must be in the same subnet as that of the ISU3. This can be achieved by one of two methods. Which method is employed can be influenced by your experience level, but is ultimately dependent on what the target configuration is (see Network Config. Options) and your desire to reduce the setup steps involved.

The first method is to set the IP address of the PC so that it is in the same range (subnet) as the ISU3 default address (See Appendices : PC IP Address Setting). The extra work involved in this method is that if your PC, or laptop, does not usually have this address you will need to set it back to normal afterward; you may also need to re-set the IP if the ISU3 address is changed in the setup process, so that you can maintain programming contact with it.

The second method is to use a serial cable to connect from the PC to the ISU3 Console port, and then use the console interface of the ISU3 to set the IP address of the ISU3 to the desired address initially. Then the PC can be connected to the ISU as above for programming setup with generally no other IP changes necessary. This second method is also useful if the ISU3 IP address is other than default and/or unknown initially. For details of this method see Appendices : ISU3 Console Methods.

The factory default Ethernet address of the ISU3 is **http://10.10.10.6:30061**
With PC connected as above, and presuming the ISU3 is at default IP, launch your Web Browser (Internet Explorer, Firefox, etc) and type the above address into the address bar. If your connections and device addresses are correct you will get the ISU3 login dialog. Viz:

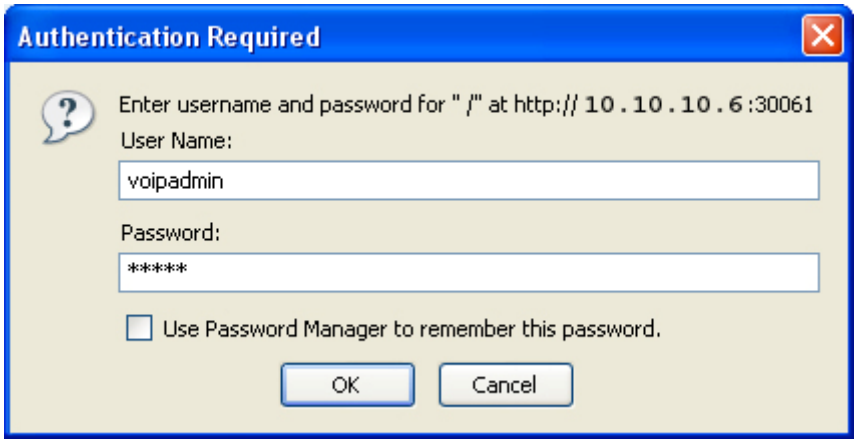

Fig. 1.6
ISU3
Login Dialog

Enter the default username : **voipadmin** and password : **admin** .
Then click **[OK].**

After Login (and sometimes a short wait) the first page presented will be the Information Page. Viz:

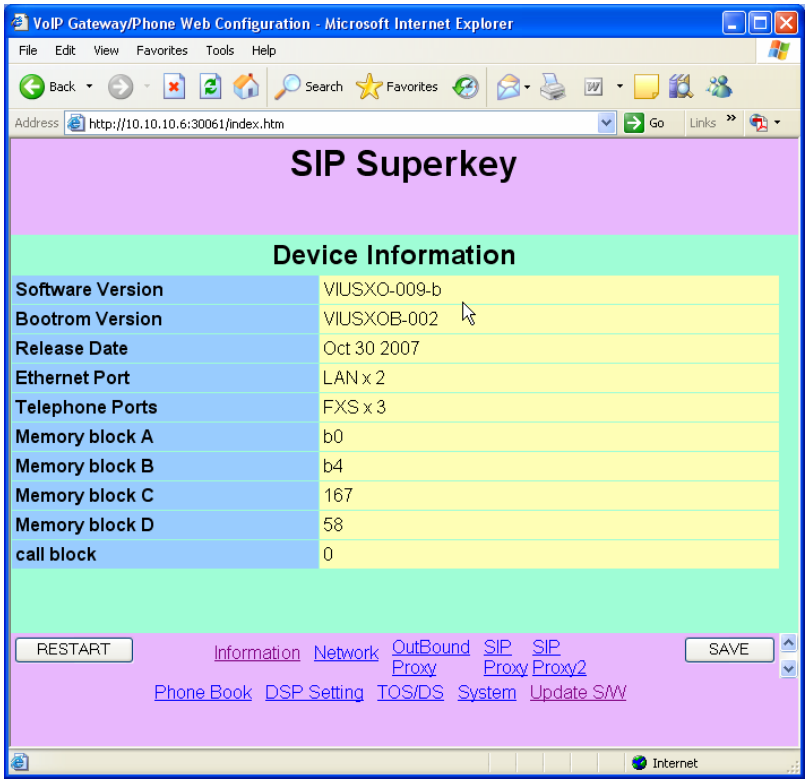

Fig. 1.7 Information Page

The information items of note on this page are the Software Version, and Bootrom Version. The rest of the ISU3 web interface is navigated using the page links in the lower section.

Network Settings

Presuming setup from default - the first settings to be made will be the network details. Click on the “Network” link and the following page will be shown :

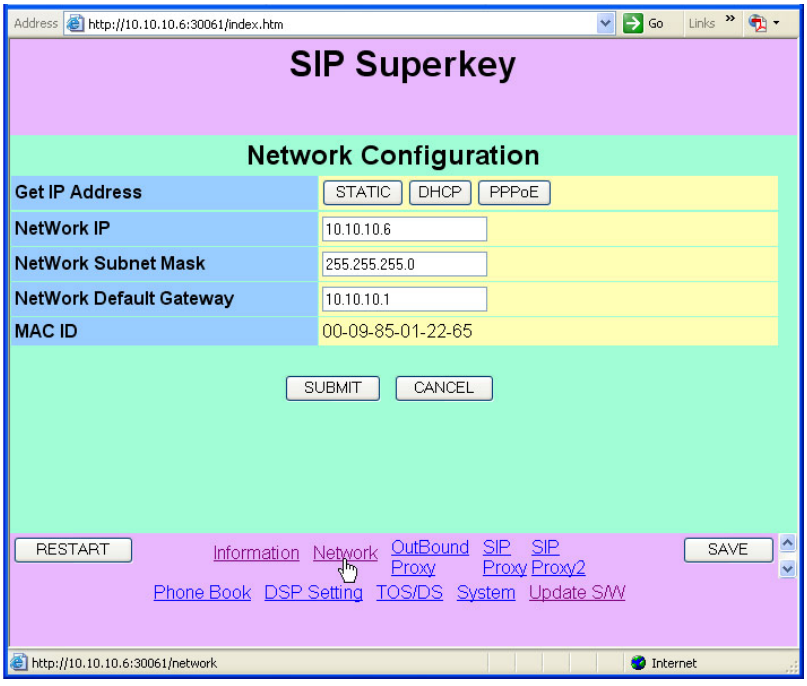

Fig. 1.8 Network Configuration Page

The default Network type is “Static”.
The settings to be made (if not already done so by the Console method) are :
Network IP (the IP address of the ISU3 itself), Subnet Mask (of the network or subnet), and Network Gateway (the IP address of the network border router).
For network configurations of types 1 & 2 (LAN and NAT’d) these will be private IP addresses and mask for the LAN the ISU is to be connected to. The same applies to network type 4 (Public IP) with the exception that real IP’s and more restricted subnet mask will be used.
If you are in control and setting up a dedicated connection these will be known. If you are connecting to a business network (or VPN) you will need to obtain an IP address (and other details) to use from the network administrator
When the desired settings are made - remember to click [Submit] before leaving the page.

PPPoE Network Setting

If the network configuration to be used is PPPoE - click on the [PPPoE] button and the following page change will be displayed :

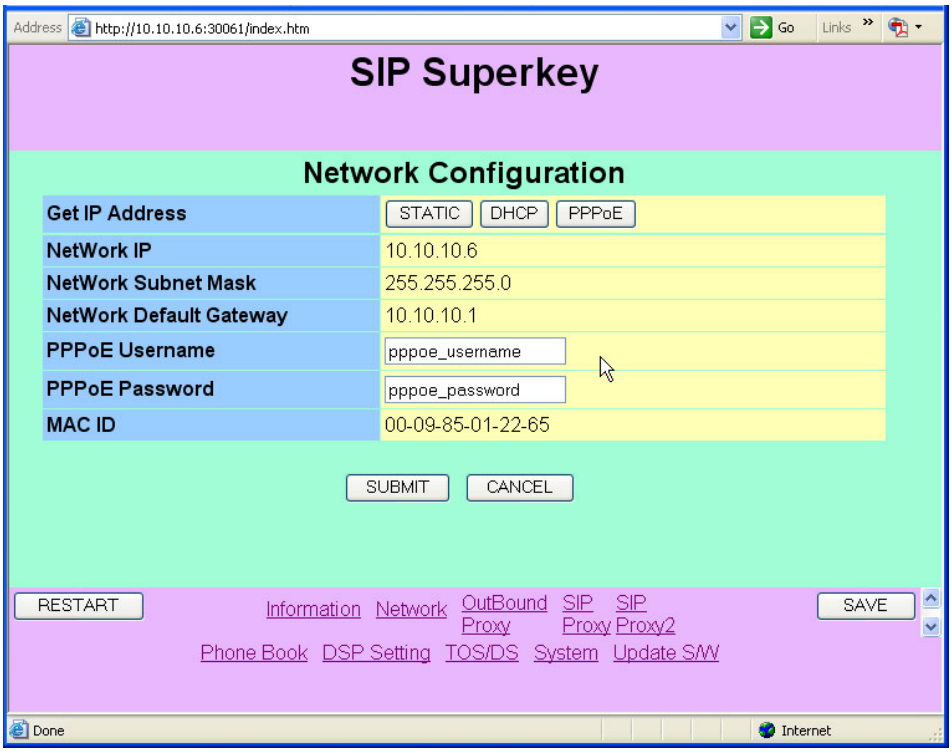

Fig. 1.9 Network Config. PPPoE

The settings to be made here are the PPPoE username and password.
Since PPPoE mode is usually a direct connection (via a bridge-mode modem) to the ISP’s broadband feed, or “account” as it is generally known, these are the username and password necessary to connect to the “account” and will be supplied by the ISP.
After setting these remember to click [Submit] before leaving the page.

NB: If a mistake is made in these settings preventing a login to the ISP then the Console method must be used to issue the command “set dhcp off” and then set an IP (eim ip) before the Web interface of the card can be regained to correct the error. See Appendices :Console.

The Network IP, Mask, and Gateway fields will show the real IP etc of the account (and therefore the ISU3 IP) after the PPPoE login is successful as they are supplied by the ISP equipment as part of the login process. Since the (ISU) requirement is for a “static” IP this will usually be known beforehand. If not it can be discovered via the Console method (eim ip) after login. The IP will need to be known to browse to the ISU3 at any later stage, moreover it will be used as the “Registrar”, and usually “Proxy”, address to be set in any SIP endpoint device used as an extension.

OutBound Proxy

Click on the "OutBound Proxy" link and the OutBound page will be displayed :

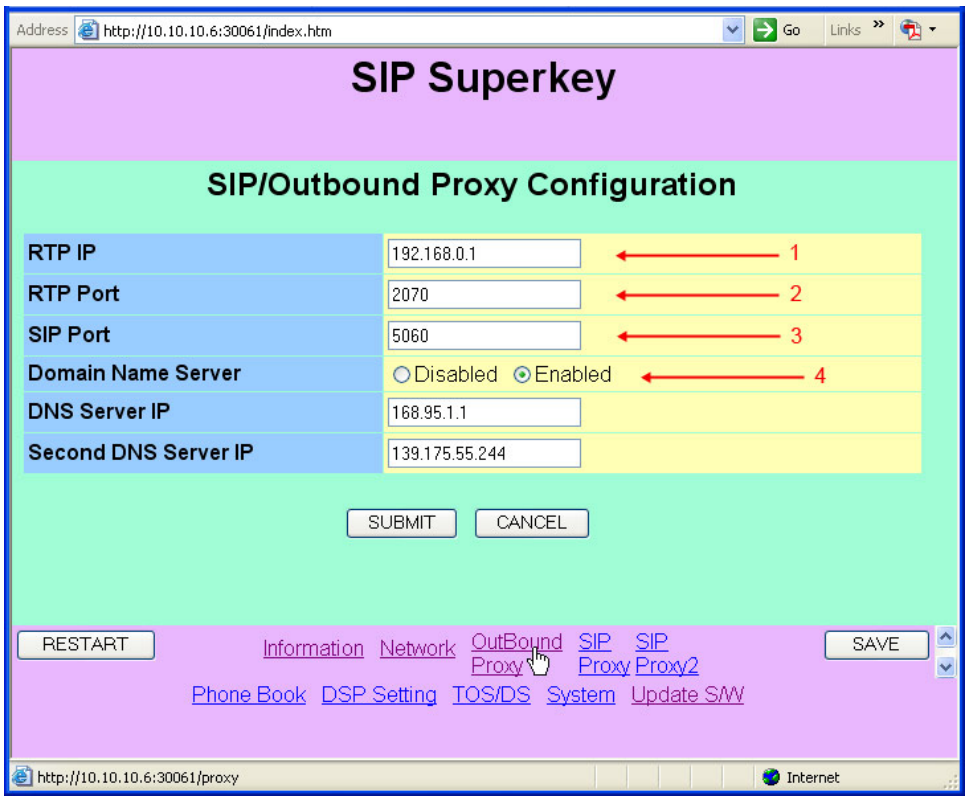

Fig. 10 OutBound Proxy Page

The OutBound Proxy page is not quite as it sounds.
On the ISU3 this page determines the traffic portal for traffic of both directions but more importantly incoming as the settings here determine the network identity of the ISU3 with respect to voice connections from the SIP endpoint devices being used as extensions.
The mandatory settings are the RTP IP, RTP Port, and SIP Port. Remember to [Submit].

1. **RTP IP :**
   For all network types (configurations) except type 2 (NAT'd) this will be set to the IP address of the ISU3 itself. For the PPPoE type this may need to be set after the inherent real IP address granted by the ISP is discovered.
   For a **NAT'd** network this IP address will be set to the **WAN IP** address of the router the ISU is situated behind. The router will then be set to "Port Forward" the ports described below to the LAN IP address of the ISU3.
2. **RTP Port :**
   This will be set to the start port of the port range used to transport incoming RTP voice connections. The port range to use is recommended at 100 ports so the "Port Forward" range to set on the router - for the default 2070 shown above - will be 2070 to 2169 UDP. If multiple ISU3's are set up in the same NAT'd network : successive ISU start port (and ranges) must be staggered to prevent collision. For example ISU3-1 at 2070 to 2169, ISU3-2 at 2170 to 2269, etc.
3. **SIP Port :**
   For the ability to setup multiple ISU's the SIP port is also configurable here.
   In the same way as (2): ISU3-1 set to 5060, ISU3-2 set to 5061, etc. all UDP. This is only a single "Port Forward" to the ISU3 LAN IP to set on the router, for and to each ISU3 installed. NB: this port setting must also agree with set on the SIP Proxy page.
4. **Domain Name Server:**
   This setting is optional. It is not really necessary because the usual setup of the ISU3 is done with IP addresses, and not full domain names that need resolving, so disable. The exception to this would be if DynDNS services usage was forced - see Appendices.

SIP Proxy Settings

Click on the link for “SIP Proxy” and the main SIP Proxy page will be displayed :

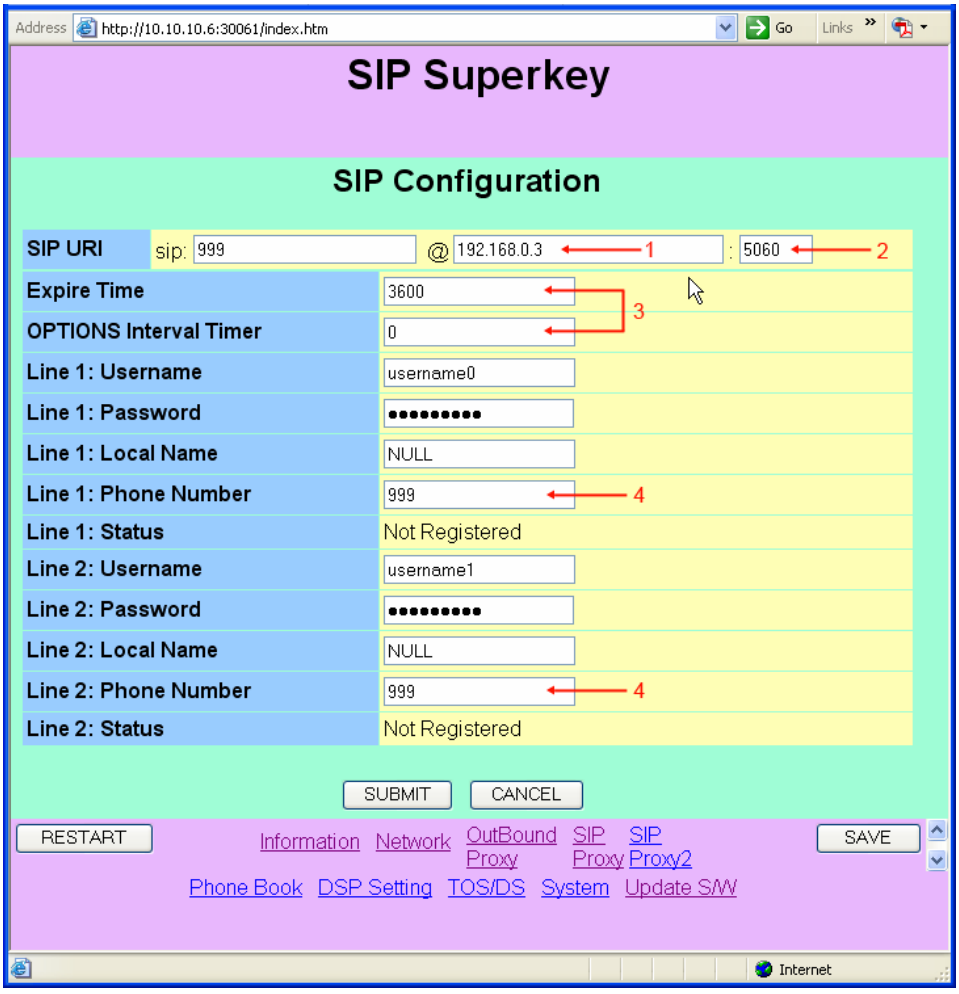

Fig. 1.11 SIP Proxy Page

The SIP Proxy page determines the SIP “Registrar” settings for the ISU3 among other things.

1. **SIP URI address :**
   This is the “SIP Registrar” address.
   This is set to the same address as the RTP IP address on the OutBound Proxy page.
   It is also the address that will be used as the “SIP Registrar” parameter for the programming of the SIP endpoint devices set up as IP extensions.
2. **SIP URI Port :**
   This is the “SIP Registrar” port.
   Set to the same value as the SIP Port on the OutBound Proxy page.
   As per the notes on the OutBound Proxy page this port value may vary where more than one ISU3 is installed. As per (1) above this is also the “SIP Registrar Port” parameter for the SIP endpoint device/s programming.
3. **Expire Time and Options Interval Timer :**
   Sets the registration interval (seconds) for SIP endpoint devices. Set both these fields to a value less than the expected port binding time of the router/s the SIP endpoints will be situated behind. Just under 3 minutes (180 sec) is usually acceptable. The endpoint devices are then instructed to maintain this interval or less, so maintaining contact with the ISU.
4. **Phone Numbers :** On the ISU3 these are a porting number value that the GDS uses for control. They will usually be left at the default value (999).
   However the rule is : no other extension in the GDS system must be given the same number. So if extension number 999 already exists on the GDS, either the ext. No. or the “Phone Numbers” must be changed or faulty operation will result.
   This also applies to the “Phone Number” on the SIP Proxy2 page.

DSP Settings :

Click on the "DSP Setting" link and the DSP Settings page will be displayed :

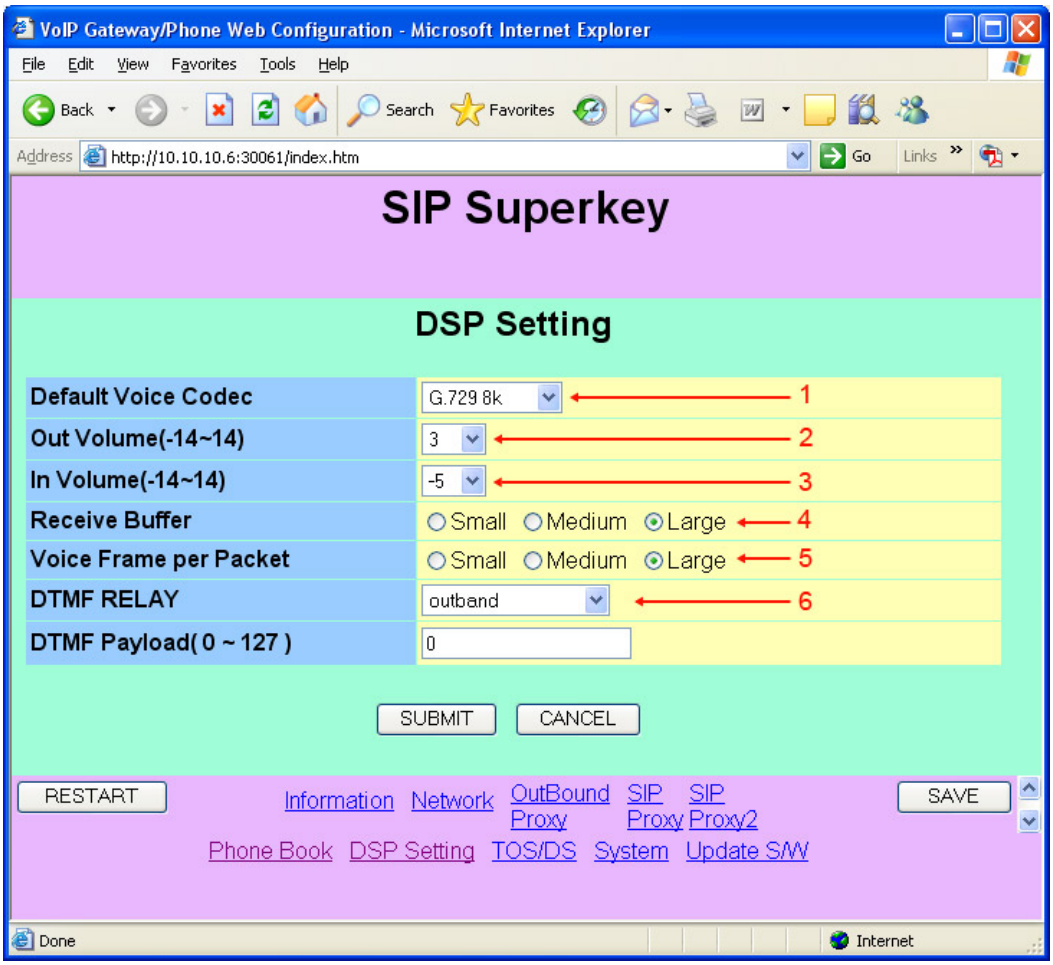

Fig. 1.12 DSP Settings Page

The DSP Settings page determines the parameters of the voice link.
The above graphic shows the preferred settings, not the default page.

1. **Default Voice Codec:**
   Whilst the ISU3 is auto sensing with regard to voice compression codec in use, the default (highest priority) codec needs to be specified. (For more information on codecs see appendices :Voice Codecs). The recommended codec for general use is G729. This setting is made by selecting from the drop menu shown by clicking the ellipsis (down arrowhead) at the right of the value field.
2. **Out Volume:**
   Determines the voice volume transmitted from the ISU to the SIP extension devices. A setting of between 5 and 10 has been found to give acceptable levels for send volume. Set by select from the ellipsis drop menu. Can be adjusted to taste.
3. **In Volume :**
   Determines incoming volume translation of voice data stream from SIP extension devices. Like a gain adjustment, adjust to 0 (or taste) by selection from the ellipsis drop menu.
4. **Receive Buffer :**
   Determines the size of the buffer used for incoming voice packets. Both this and the following item will follow a default setting when the codec setting is submitted. Can be adjusted according to network needs. Trade-off is: the larger the buffer the more tolerant of network voice packet transit time differences but the higher the voice delay incoming from SIP endpoint to the system.

5. **Voice Frame per Packet:**
As with the previous item this value will follow a default setting when the voice codec setting is submitted. Relative to the Small/Med/Large settings : actual milliseconds of voice per packet varies according to codec. Can be adjusted - but test voice if altered. Trade-off is : the larger the voice frame size the less packets are sent to network but the quicker the voice quality will drop off if packets are lost on the network.
Do not use the "Medium" setting when using G729 codec - one way voice may result.

6. **DTMF Relay :**
This parameter determines how post dial DTMF is sent across the network.
From the ellipsis drop menu it can be seen there are three options :
"Disable" means DTMF will be sent as tones within the voice (audio) band.
This option suffers markedly when higher compression codecs are used (G729 and particularly G723). In effect the DTMF tones get mangled by the compression process and any packet loss exacerbates the problem making the DTMF hard to detect at the receiving end. Currently GDS functions will not respond to commands using "Disable".
"Inband(RFC2833)" is not functional in the code set of the VIU/ISU3 card - don't use.
"Outband" means the DTMF digits will be sent as an information packet in the SIP command stream. Known as "SIP Info" style DTMF, this is the most reliable available and recommended for use. When the Outband setting is chosen SIP endpoint devices used as connected extensions also need to be programmed for "SIP Info" DTMF.

The last field "Payload" is related only to RFC2833 DTMF and therefore not relevant.

TOS/DS Settings

Click on the "TOS/DS" link and the Type of Service page will be displayed :

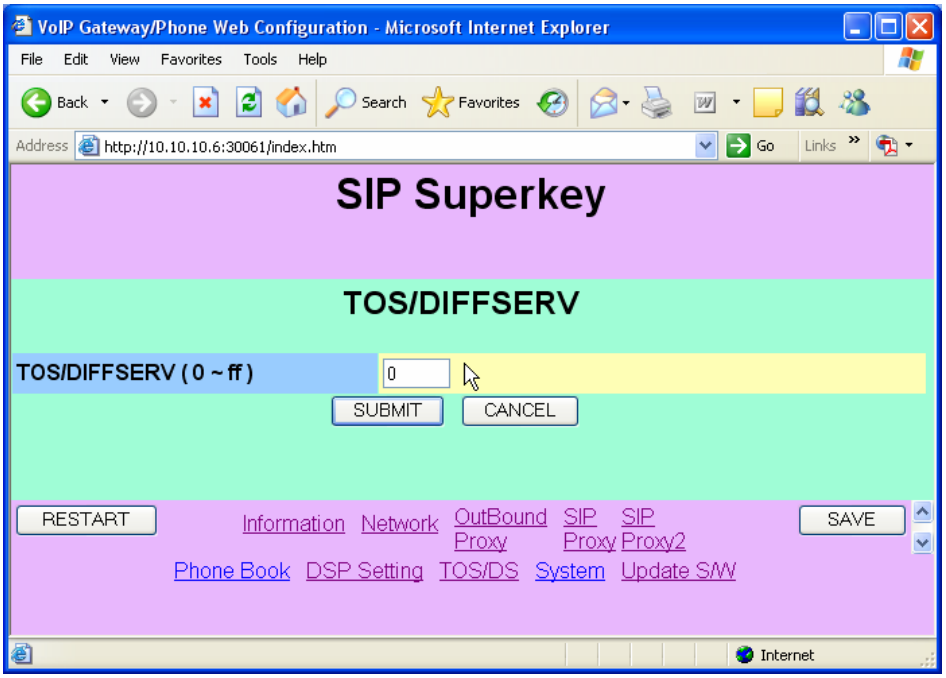

Fig. 1.13 TOS/DS Page

Type of Service / Diffserv relates to Type of Service (TOS) or Differentiated Services Code Point (DSCP) which is a method of packet marking that allows enabled network routers to prioritise the handling and passage of packets according to their marked value. Appropriately marked packets, in this case VoIP packets, are thus afforded priority which may reduce transit times and/or packet loss and thus alleviate voice quality problems from these sources. The use of, and value set, of this parameter will be dictated by instruction, by network system administration, or ISP support. Since our recommendation is dedicated network porting for the ISU and its voice traffic, these settings are most relative to the ISP supporting this connection. This represents an option that can and should be taken up if it is supported by your ISP. Note: the value set in this field is entered in hexadecimal form and constitutes the entire TOS/DSCP octet value.

System Settings

Click on the “System” link and the System Settings page will be displayed :

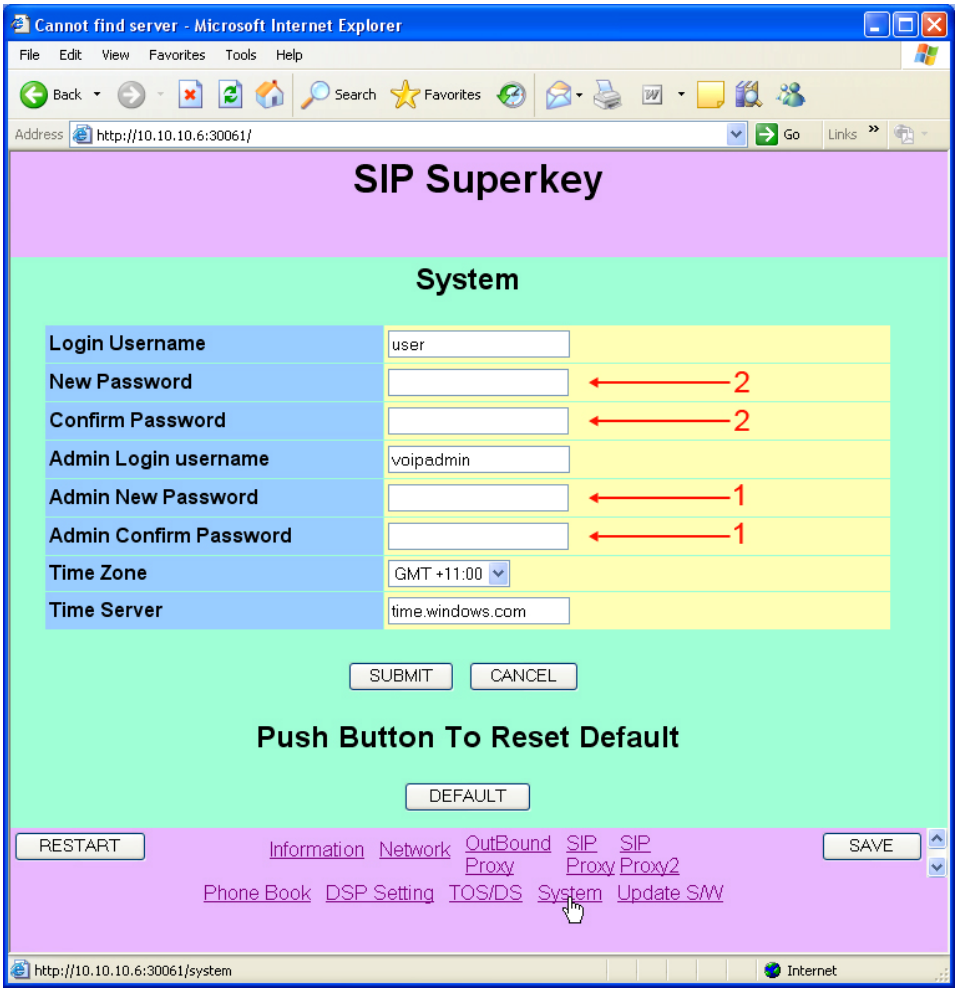

Fig. 1.14 System Page

The System page allows the setting of login usernames and passwords.

There are two login qualifications:

- User login allows edit of the Network Settings page and the User details of the System page. All other pages and settings can be only be viewed.
- Admin login allows “access all areas” - all interface pages can be edited.

Because the ISU3 is, by its nature as a Registrar, available either directly or ported from a real Internet address it can be accessed by browser from any Internet access point. For reasons of security then it is mandatory that a strong password be set, for both the Admin and User login authentications to prevent unauthorised access.
A strong password is any non-dictionary combination of both letters and numbers of at least eight characters in length.
Since the User login doesn’t allow much to be edited it might seem safe to leave this at default. IT IS NOT ! If the string “set default factory” is entered in the Login Username field on this page and submitted: the card will immediately reset to factory default settings.

1 & 2 **Enter appropriate passwords for both User and Admin logins.**
Click the [Submit] button to save the settings before leaving the page.
Make a note of the passwords set in your install notes !!

The Time Zone and Server settings are not of any consequence for this implementation.

Phone Book

Click on the “Phone Book” link and the Phone Book page will be displayed :

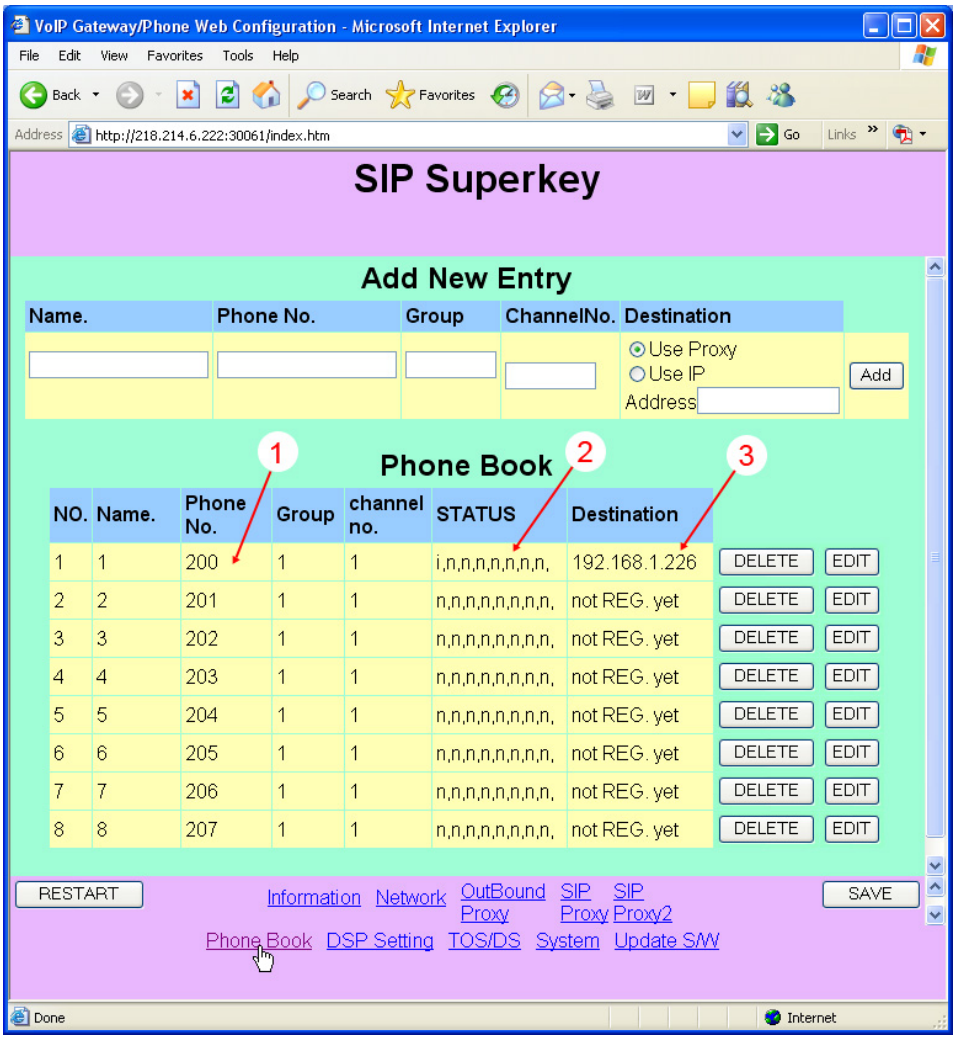

Fig. 1.15 Phone Book page

Unlike other implementations built on the VIU hardware platform the ISU3 Phone Book page is not editable. What the ISU3 Phone Book page shows is the status of the IP extension settings and devices :

1. **Phone No. :**
   The Phone No. column shows the extension numbers allocated to available positions (ports) in GDS programming Mode 21 - see Fig 4 (pp8). Ext No’s shown are example only As noted elsewhere: the ISU3, when mounted in any GDS trunk slot has eight ports available for IP extension assignment and therefore device registration. The exception is when the ISU3 is mounted in slot 9 where the following virtual slots “a” and “b” can also be used for IP extension assignment/registration - provided no ISDN PRI card is fitted in the same cabinet. If this option is taken up these extra extension positions will also show in the above phone book.
   Please note: The mentioned GDS virtual slots xa & xb are used for ISDN PRI or ISU3 use only (mutually exclusive) - no other use is supported.
2. **Status :**
   The Status field shows the dynamic status of connected IP extension devices.
   Only the first position is used with i=idle, r=ringing, t=talking, n=not registered.
3. **Destination :**
   The Destination field shows the real or virtual IP address of devices that are registered (and therefore available) or “not REG. yet” (not available) as the case may be.

All other fields show standard listing details as shown. The “Add New Entry” block, and Edit/Delete buttons, are inoperative in this implementation.

Section 4

Other SIP Devices

Contents

Other SIP Endpoint Device Options and Setups

Since the purpose of the ISU3 implementation is to form IP remote extension capability for GDS systems, it would be imagined that any device that conformed to the SIP protocol (RFC 3261 et al) could be used as a remote endpoint device. Unfortunately this is not the case. Due to differences in protocol implementation by various manufacturers not all SIP equipment is interoperable. For this reason the following guide has been prepared detailing equipment that has been tested against the ISU3 and found functional, and the setup details of such.
It must be remembered from the outset that :

- Apart from the superceded Hybrex HP300 IP Phone, and the soon to be released new Hybrex IP-3861 SIP phone (which will display DSS/BLF etc similarly to a digital system handset) most if not all other endpoint devices will provide SLT (analogue extension) functionality in use. This includes call transfer, all other system SLT features such as call forward etc, including access to system voicemail (if fitted in the parent system).
- Although the ISU3 can register commonly eight, but up to twenty-four (conditional), extension devices the ISU3 hardware base only supports three concurrent active calls. Unlike the Hybrex SPU90 proxy server the ISU3 does not proxy calls between remote extensions, each active call is a call into the parent GDS just like any other system extension. This allows call handling in a true “extension” sense with all of the inherent advantages of such.

SIP Handsets:

SNOM Series:

The European manufacturer Snom produces a range of SIP phones which have a very elegant European style and extensive programmability. The most obvious handset for use with the ISU3 is the Snom300, with the Snom320 also open to consideration. The 320 however may be a little underutilised considering it’s capabilities if it is only used for ISU3 work.

Snom 300

Snom320

Fig. 4.1
Snom Phones

Full Details of these phones can be found here: http://www.snom.com/en/products.html

The Snom phones are fairly easy to set up. They are by default set to obtain an IP address automatically (DHCP) so after unpacking and connecting to power and network (LAN) they are ready to program. Some basic settings can be made from the keypad of the phone itself, however to set them up the way we need initially we need to use their Web interface. To do this it is necessary to discover the IP address the phone has obtained which can be done using the manual excerpt shown at right.
Then open a browser (IExplorer or other) on a PC connected to the same network as the phone and enter the phones IP in the address field : for example http://192.168.1.10, then tap the Enter key. The various web pages obtained and the settings to make are shown following.

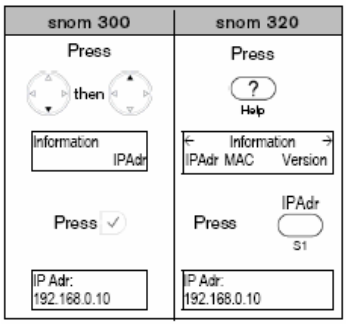

Fig. 4.2
Snom IP

Snom Index Screen

Below is the first screen shown when logging onto the Snom phone Web interface.
The following setup screens shown are those of a Snom320 which will be seen to be a very programmable device. The more applicable Snom300 phone has the same type of Web interface with the exception that some features are reduced - eg fewer “Identites” and fewer “Function Keys” -apart from those the setups are the same. Phone IP shown is example only.

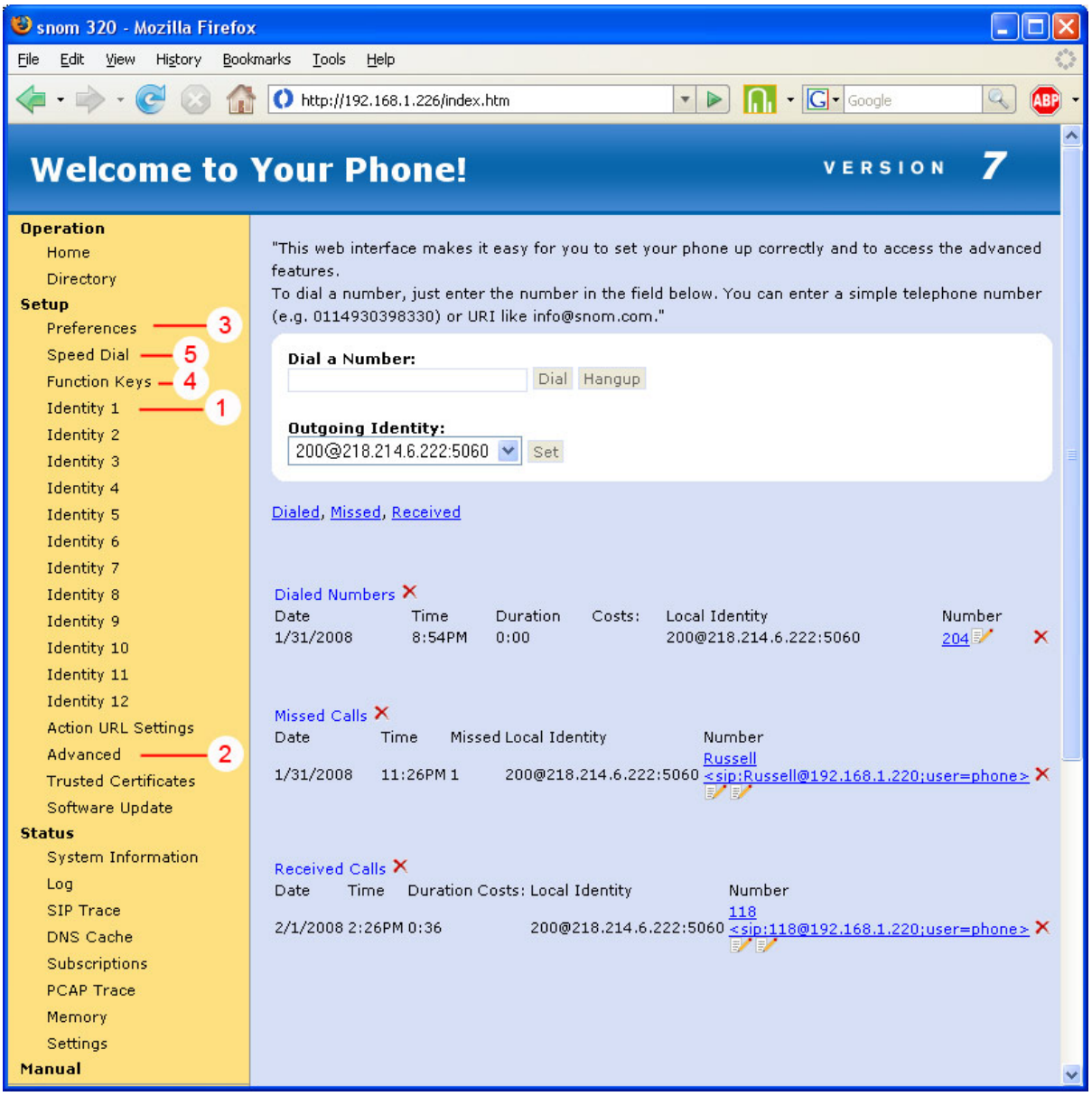

Fig. 4.3 Snom index screen

As can be seen the index page shows call logs and allows dialing PSTN No.s direct from the page from the “Dialed Numbers” list only. This is due to the prepended 9 necessary for system trunk access. However dialing by click for extension call records works from all lists.
Your attention is now called to the menu on the left hand side.
The menu items required to be set up are tagged in level of importance and as a guide to the following page sequences. Other settings can be made but the important ones for operability are shown:

1. **Identity** : where the relevant settings are made for each “identity” set up for the phone.
2. **Advanced** : where setups are made that govern the operability of the phone as a whole.
3. **Preferences** : where setups are made that affect the operation in a preferential manner.
4. **Function Keys** : where functions of the phone’s programmable keys can be re-assigned.
5. **Speed Dial** : for the entry of personal speed dials.

Snom Identity Screen - Login Tab :

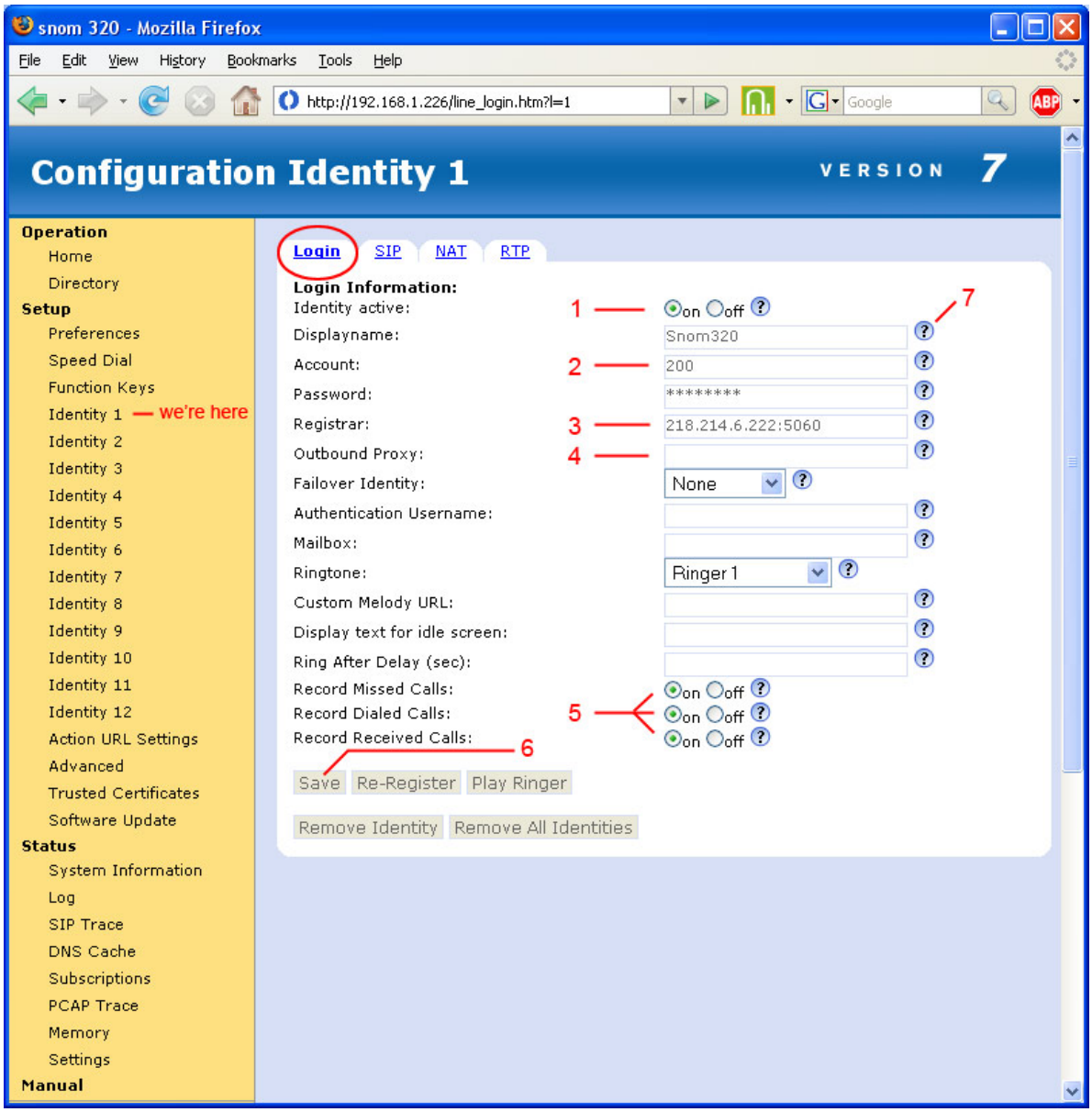

Fig. 4.4 Snom Identity - Login Tab screen

The Login Tab is the default entry point for any Identity:

1. **Identity Active** : On by default. On = Identity will register with specified server. If a particular Identity isn't required for normal use then set to Off - will reduce traffic.
2. **Account** : Is set to the SIP Account Number - for now this is the Extension Number allocated in host GDS programming (one with ISU3) for this IP extension phone.
3. **Registrar** : Is set to the WAN IP and port of the ISU3 card - see ISU3 setup section 1.
4. **Outbound Proxy** : Not generally used for this implementation - may be used according to Network Admin instruction (special cases).
5. **Call Records** : Set according to preference - default is On.
6. **Save** : Remember to Save any settings made before leaving the page/tab. NB: when settings are changed there will often be a prompt to "Reboot" the phone - allowing settings to be applied. This is only necessary after all setting changes have been made.
7. You will notice the question prompt on all setting parameters - clicking this icon will take you direct to an Internet Wiki for Snom features (Internet access required). This is useful for explanations but by no means absolutely comprehensive. As noted prior - other settings can be made if desired (or for experiment) - the main ones for this implementation are detailed.

Snom Identity Screen - SIP Tab :

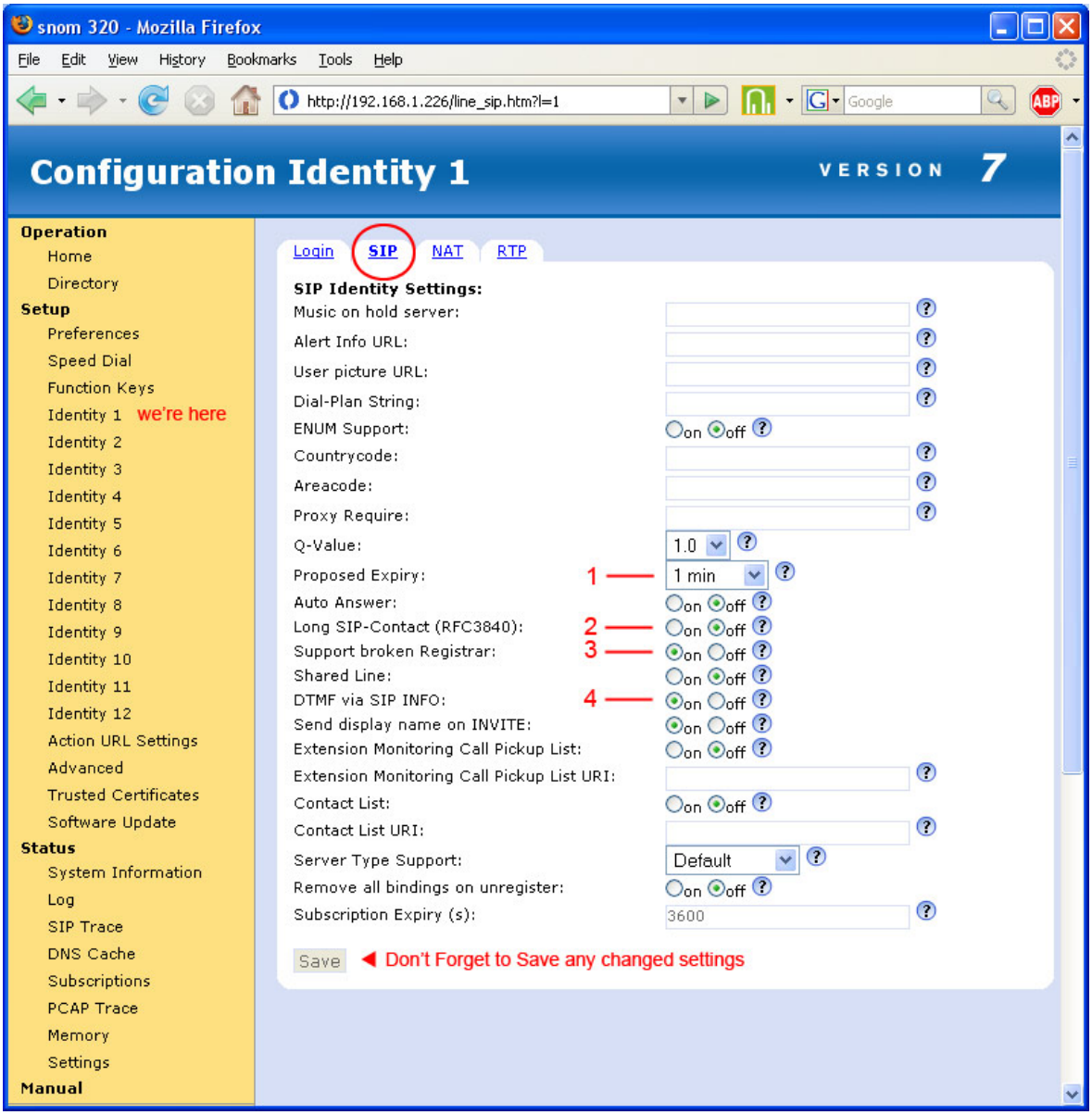

Fig. 4.5 Snom Identity - SIP tab.

Generally the setup shown can be used in it's entirety as this is functional.
As noted before other settings could be made if you are brave or knowledgeable but no guarantee is given of operability other than stated.

The important settings are shown :

1. **Proposed Expiry** : Set to 1 minute - this sets the registration interval and therefore the messaging that keeps the port bindings on the NAT router this phone is likely behind. These port bindings are what enable the phone to receive calls from the ISU3.
2. **Long SIP Contact** : Set to OFF - this is the verbose form of Contact annunciation in the SIP protocol - ISU3 doesn't use it and won't register phone if it is used.
3. **Support Broken Registrar** : Set to ON - Phone is designed for use with complex SIP servers - ISU3 is simpler by comparison - Phone won't accept ISU3 register response if this parameter is set to off.
4. **DTMF via SIP Info** : Set to ON - post dial DTMF is sent via SIP Info method which is reliable and ISU3 supports. In fact ISU3 will only accept SIP Info post dial DTMF.
   NB: there is an issue with Snom DTMF that is in the process of rectification so this page may change - see notes at end of this section.

Snom Identity Screen - RTP Tab :

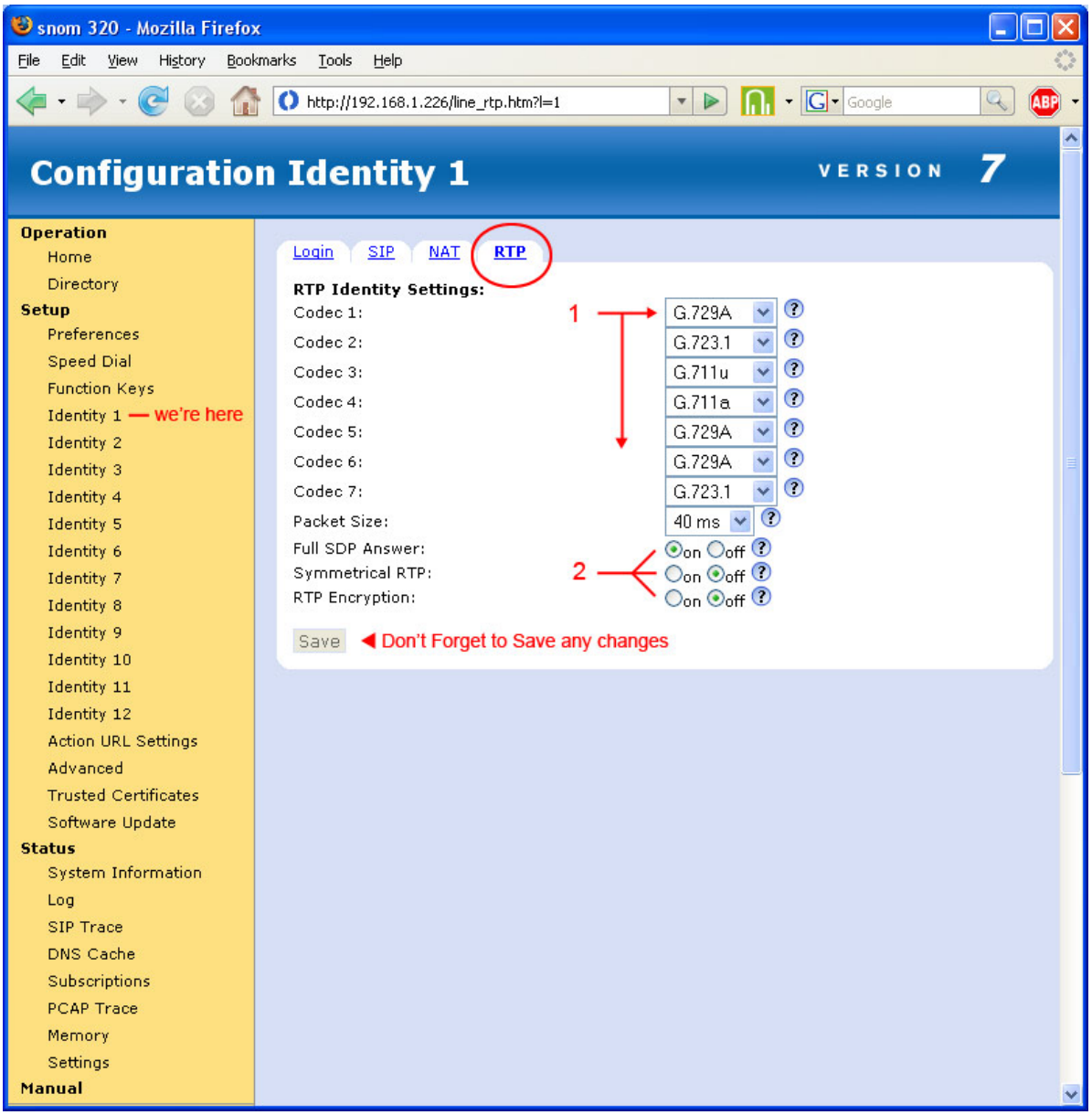

Fig. 4.6 Snom Identity - RTP Tab.

The Identity - RTP Tab allows the setting of the Codec priority and other RTP protocol parameters for this identity :

1. Set the required Codec type for your planned implementation. See Appendices : Codecs The codec order set will be used according to order of priority, and availability and/or capability of the connecting equipment - for example codec "1" will be used if the distant equipment supports it, otherwise codec "2" will be used, and so-on.
2. The settings shown work. Other settings may or may not.
   RTP encryption must be set OFF as the ISU3 does not support it.

Note : The Identity - NAT tab settings are not used for this implementation
- they allow setting of STUN servers and such, commonly used for P2P and server-less connections under certain circumstances (not effective with symmetric NAT firewalls) - STUN is not used by the ISU3 as it is engineered as a registrar/server.

Snom Advanced Settings : Network Tab

Click the menu “Advanced” link for the Advanced tab set :

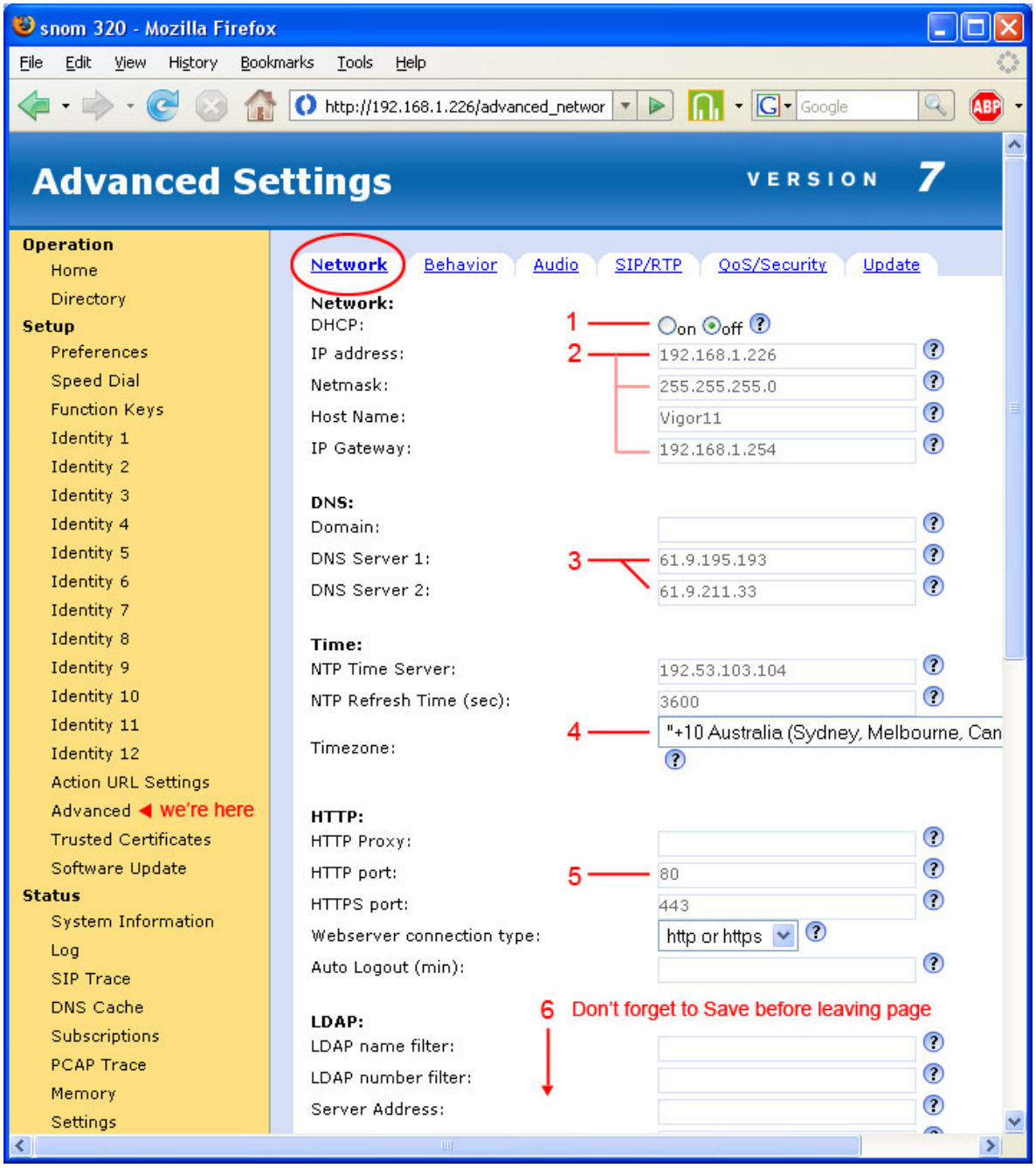

Fig. 4.7 Snom Advanced Settings - Network Tab

As with other pages the Advanced Settings will mostly be left at default.
Relevant settings changes only are shown :

1. **DHCP** : The decision to change from default of DHCP ON will be based on whether the phone will be remotely administered by port forwarding to it through the border router. If remote administer is desired the IP address of the phone must be fixed (DHCP = OFF)
2. **IP Address** : If DHCP is Off the phone must be assigned a fixed address, and this must be set to an IP outside of the routers DHCP “pool” (see network administrator for IP grant). The Subnet Mask and Gateway will already be known if the phone is already on the destination network; if not relevant values for the destination network are entered.
3. **DNS** : Enter the preferred DNS server IP’s for the ISP network the phone is connected to.
4. **Timezone**: Select the relevant timezone for your location, as time is shown on the phone
5. **HTTP port**: Change only if you have a specific plan in mind - for example you want to set up remote access to the Snom phones and the router the phones are behind cannot do port redirect, only port forward. So each phone needs to be set to a different port for web access and you will browse to them using only that port.

Snom Advanced Settings : Behavior Tab :

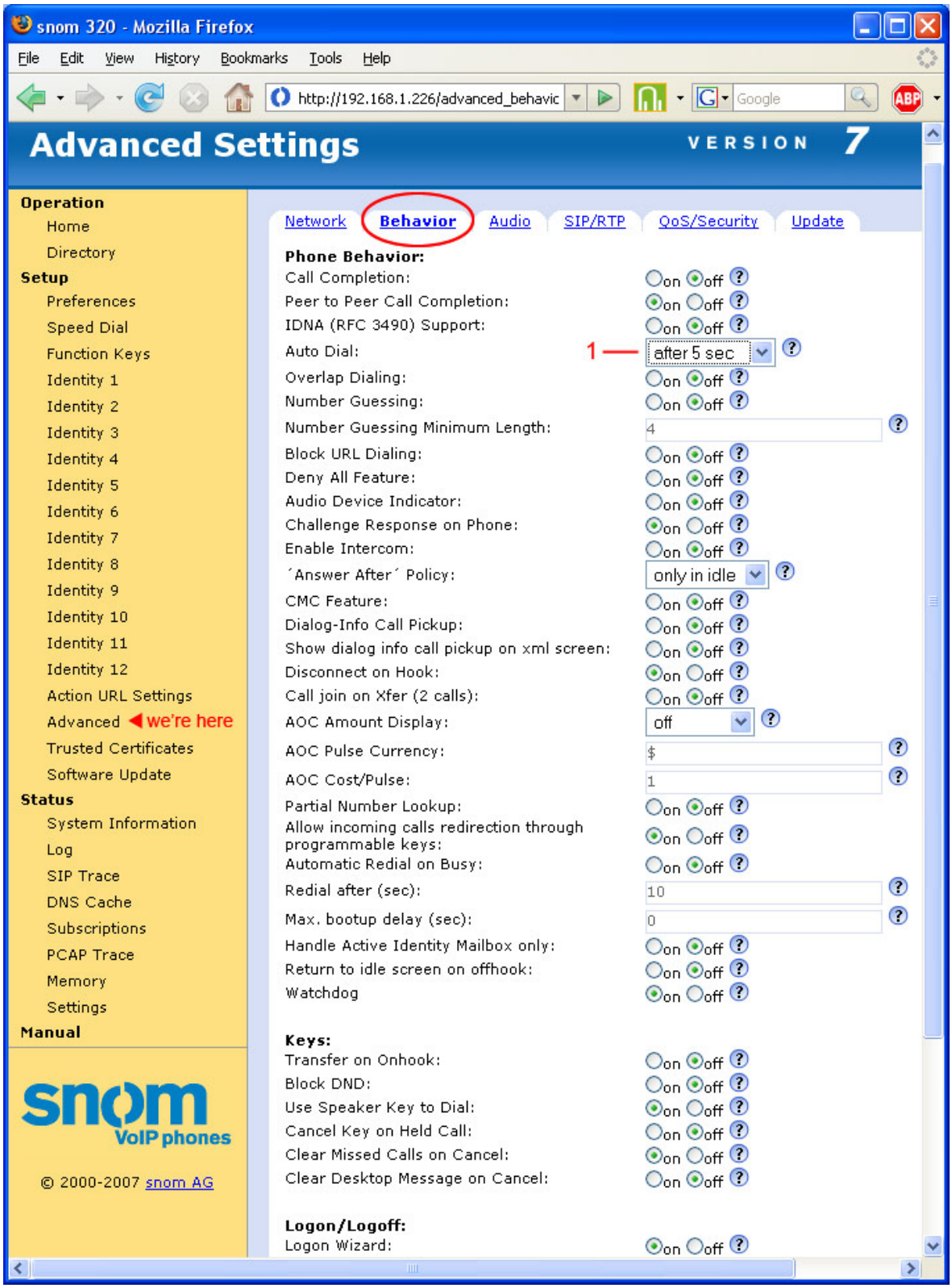

Fig. 4.8 Snom Advanced Settings - Behavior tab

The Behavior tab, as the name suggests, governs aspects of the phone's behaviour.
The options are numerous, and mostly self explanatory.
A working configuration is shown above for you to copy initially.

1. **Auto Dial** : This can be set to preference - as shown the phone will exhibit a 5 second dial timeout. This can be overridden by pressing the tick key on the phone keypad which then sends the number immediately. If set to OFF the tick key must always be pressed to dial the entered number. Remember to Save any changes.

Snom Advanced Settings : Audio Tab :

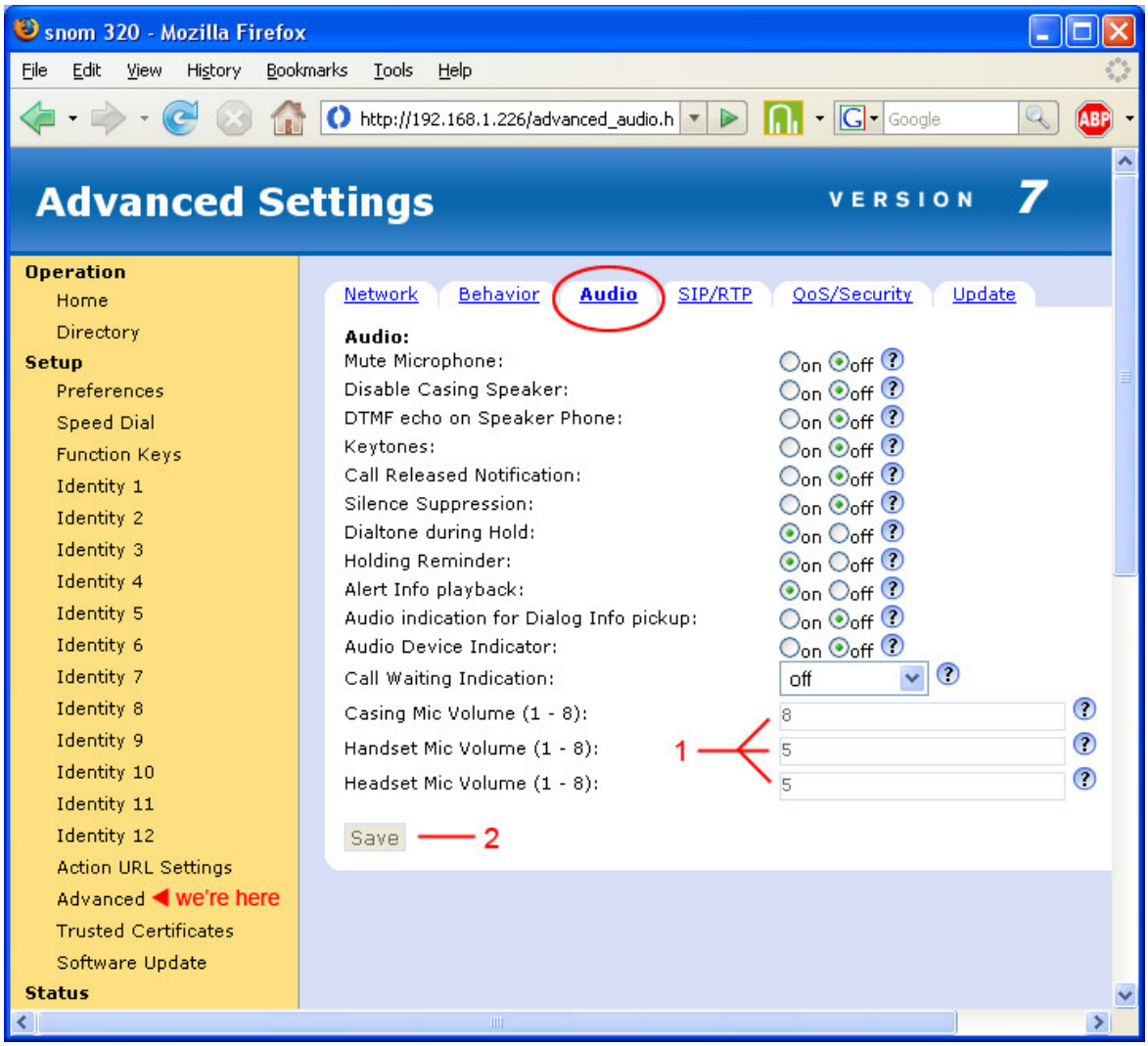

Fig. 4.9 Snom Advanced Settings - Audio tab

Working Config for general audio parameters is shown for you to copy initially.

1. **Volumes** : Are adjustable if required - suggest leave at default - otherwise exercise care if raising as audio distortion can result. Note these are Mic. (send) volumes - receive volumes can be controlled at will directly from the phone keypad (Volume button).

2. Remember to Save if any changes have been made.

Other observations:

(a) DTMF echo on speaker phone puts DTMF tones out handset speaker.

(b) Keytones means handset speaker will beep for every key press.

(c) Call released notification means handset speaker will beep loudly if distant party releases before an incoming call is picked up.

(d) Silence Suppression means less data traffic outgoing during periods of silence from the person using the phone but has the side effect of some clipping of speech and impressions of the B party that the line has gone “dead” during times the suppression is active.
These behaviours have been found to be an annoyance to some - hence the recommendation is they be set OFF.

Snom Advanced Settings : SIP/RTP Tab :

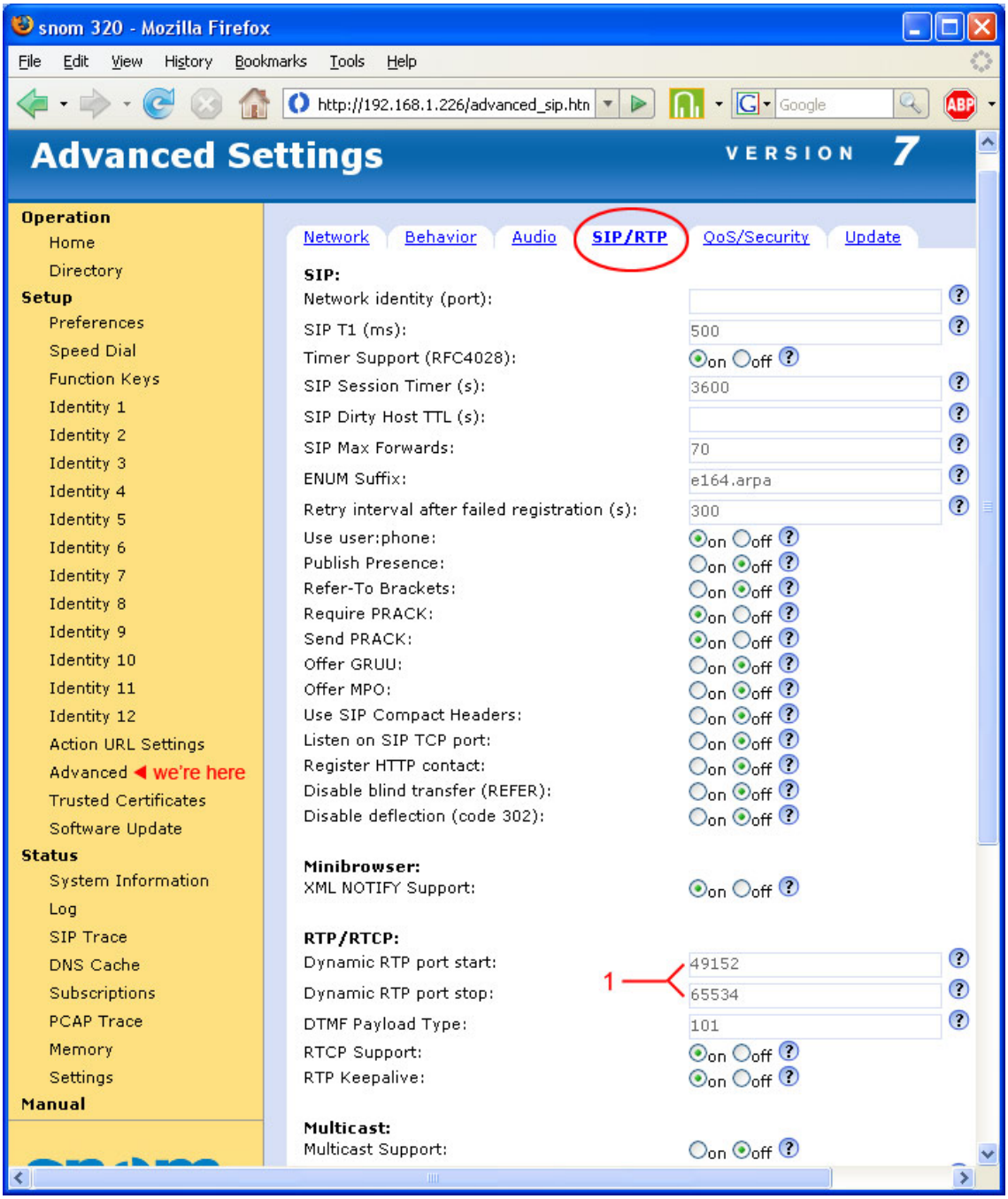

Fig. 4.10 Snom Advanced Settings - SIP/RTP

Working Config for you to copy.
Remember additional information about what each setting means can be obtained by clicking the Question mark icon at the right of each field. On a live phone Web interface and with Internet access enabled this will provide the Snom Wiki window with the term described.

1. **Dynamic RTP Ports** : This is the port range that the phone will use for dynamic assignment of its outgoing RTP ports. RTP is what carries the voice packets, and each new call gets assigned new port pairs to avoid call collision. It can be seen that the default range is very large (16382 ports). A range as large as this isn't usually necessary, a port range of 200 will suffice. It is suggested that where there are a number of these phones on the same network (LAN) that each phone be set a port range that is different to the others as a way of reducing any possible duplication/collision.

   Remember to Save any changes before leaving the page (tab).

Snom Advanced Settings : QoS / Security Tab :

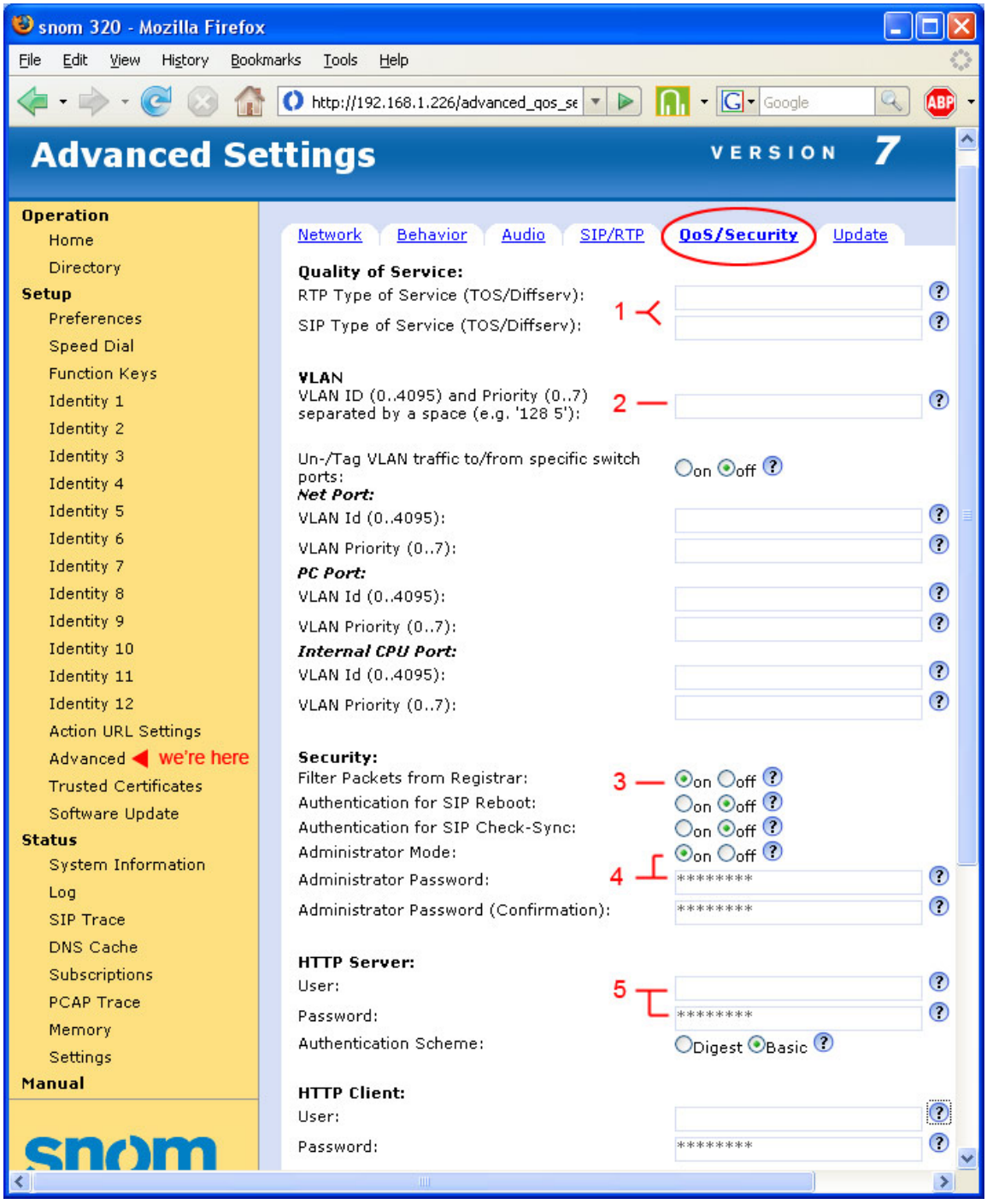

Fig. 4.11 Snom Advanced Settings - QoS/Security tab

This page allows general settings for Quality of Service and Security.
Important points are noted:

1. **ToS/Diffserv Settings** : ToS/Diffserv packet marking can achieved at the source device, and determine IP traffic priority in networks so enabled. The packet marking can be carried from source to final destination if it isn't stripped by any intermediary device, so is a very useful QoS measure. The priority scheme is programmed into the network routers therefore relevant values stated by Network Administration are entered here. The RTP streams carry voice, and the SIP streams carry command data..
2. **VLAN** : VLAN is a method for completely separating network traffic from different 'sets' of devices, and an easy way to control bandwidth allowances within the LAN/Intranet being controlled. As above, these settings, and any immediately below, are dictated by Network Administration.

Snom Advanced Menu - QoS/Security Tab (cont'd)

3. **Filter Packets from Registrar** : This means any packets not coming from the Proxy/ Registrar (in this case the ISU3) are ignored. Since this phone is being set up as a 'remote' extension off the Hybrex PBX traffic not coming from that source is not desired, hence the setting is ON. There are other observances on this concept - see Snom notes.
4. **Administrator Mode** : The screens you are seeing now and programming are "Administrator Mode" screens - like "access all areas". "User Mode" is a severely restricted mode by comparison where only a few functions like Preferences and Speed Dials etc can be set - this is desirable in most cases from the point of view of an integrator as it reduces trouble from unauthorised "play". To see what User Mode entails just set Admin Mode off and 'Save' the tab - then check around the available menu choices. You will also notice that one of the Config options available from the phone keypad is "Reset" - which is in effect a factory reset of the phone (default *all* settings). This however is only enabled after the entry of the "Administrator Password". If you have taken the step above to observe User Mode and now want to revert to Admin Mode - select the Advanced menu and enter the **Admin password - the default is 0000**.
   Given the above it is advisable to set your own Admin password - numeric digits only - (and don't forget to make a note of the password !). As a last act when setting up the phone disable Administrator mode so you can be sure any configuration changes thereafter are authorised.
5. **HTTP Server** : The meaning of HTTP Server in this sense is the phone acting as a server - i.e. giving you the web interface you are using to program. The "User" username and password you set here will be required in a login dialog before access to the phone web interface in future is granted. The factory default is blank so no login dialog is presented initially. If you want to restrict access to the web interface make the settings here. Username & password valid values: character strings, e.g. <john>, <jh24>. (Remember to make a note of any settings made !).
   The following section for HTTP client is for allowing the phone when contacting another server - say for a Config update - and is not used in the current implementation.
   Don't forget to 'Save' if any settings have been changed before leaving the tab.

Snom Advanced Settings : Update Tab :

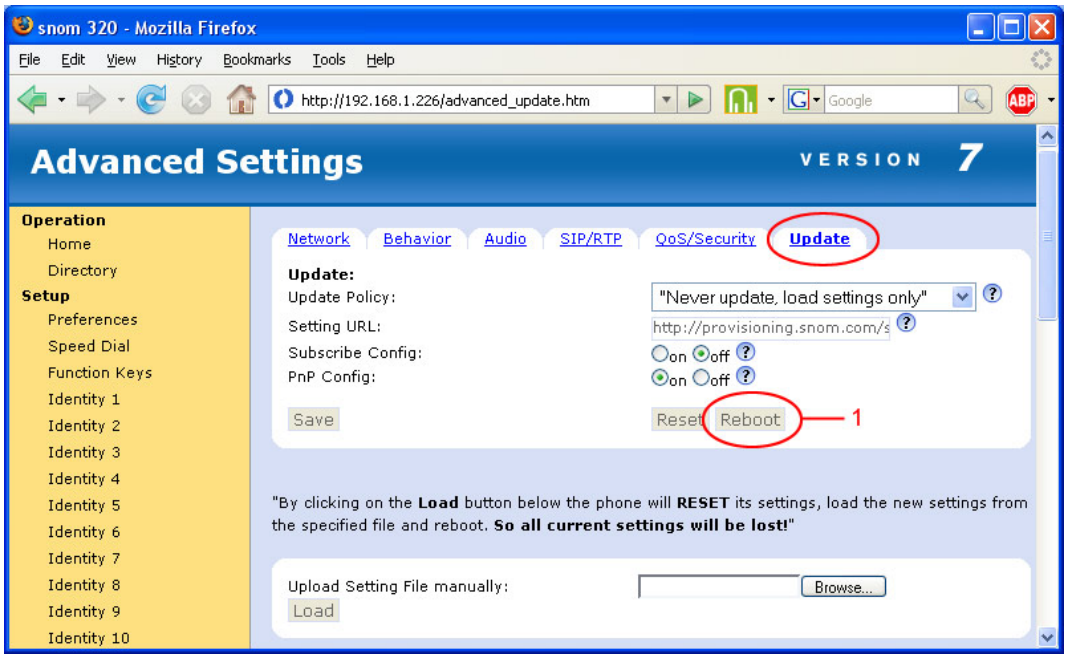

Fig. 4.12 Snom Advanced Settings - Update

The Advanced - Update tab is for "settings" downloads
These are in the form of an XML file - used for mass deployment so doesn't apply to us.
The one important thing here is :

1. **Reboot** : This is where the Web interface Reboot button is - if no other important setting changes have triggered the Snom alternate Reboot link insertion you can reboot the phone from here to be sure any changes are properly applied.

Snom Preferences Page

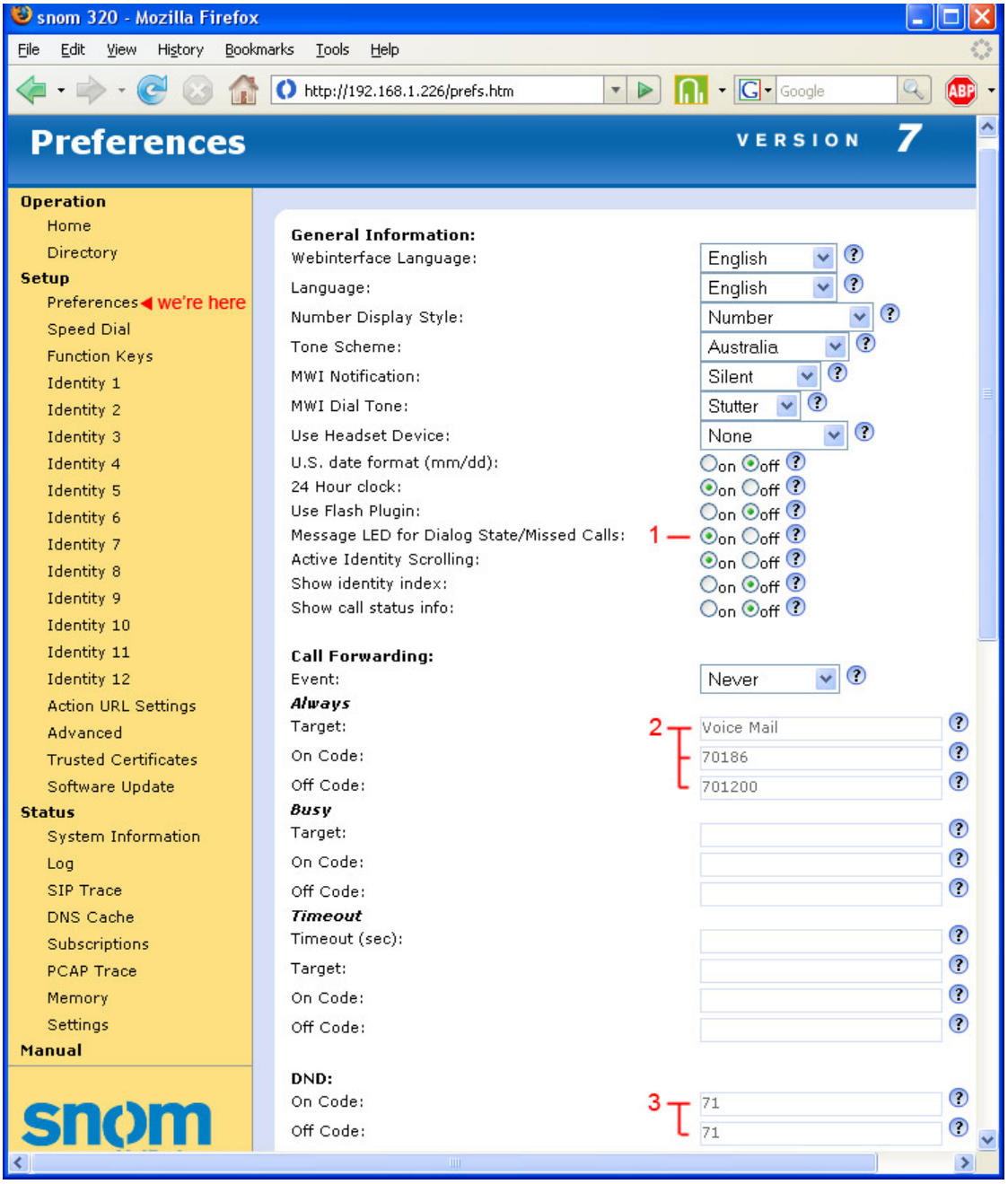

Fig. 4.13 Snom Preferences Menu - 1st half

The Snom Preferences menu is a long page so the first half is shown here. The settings shown are a guide - being open to user preference - there are some pertinent points however :

1. **Message LED** : The use of this will depend on preference and also on the settings on the Identity page > Login Tab for recording of Missed calls. See Fig 4.4 pp4-4.
2. **Call Forward Settings** : Should the call forward functions available from the Phone keypad/Nav keys be desired - enter here the Hybrex dial codes for CFwd. Shown is a common setting for ACF to Voice Mail, with the off code setting relative to this setup example extension identity of 200. The CFwd Busy, and Timeout (No Answer), settings can be made similarly.
3. **DND** : Since the parent system here is a GDS the code for DND is set in GDS programming Mode 22, and then entered here. An example is shown. This has the advantage that the Snom displays the DND status and the phone function keys can be appropriately programmed for one touch setting. The rules for the code set in Mode 22 are that it cannot begin with a * or #, and must not be duplicated anywhere else in Mode 22, extension numbering, or leading digits of PSTN numbers.

Snom Preferences Menu (second half):

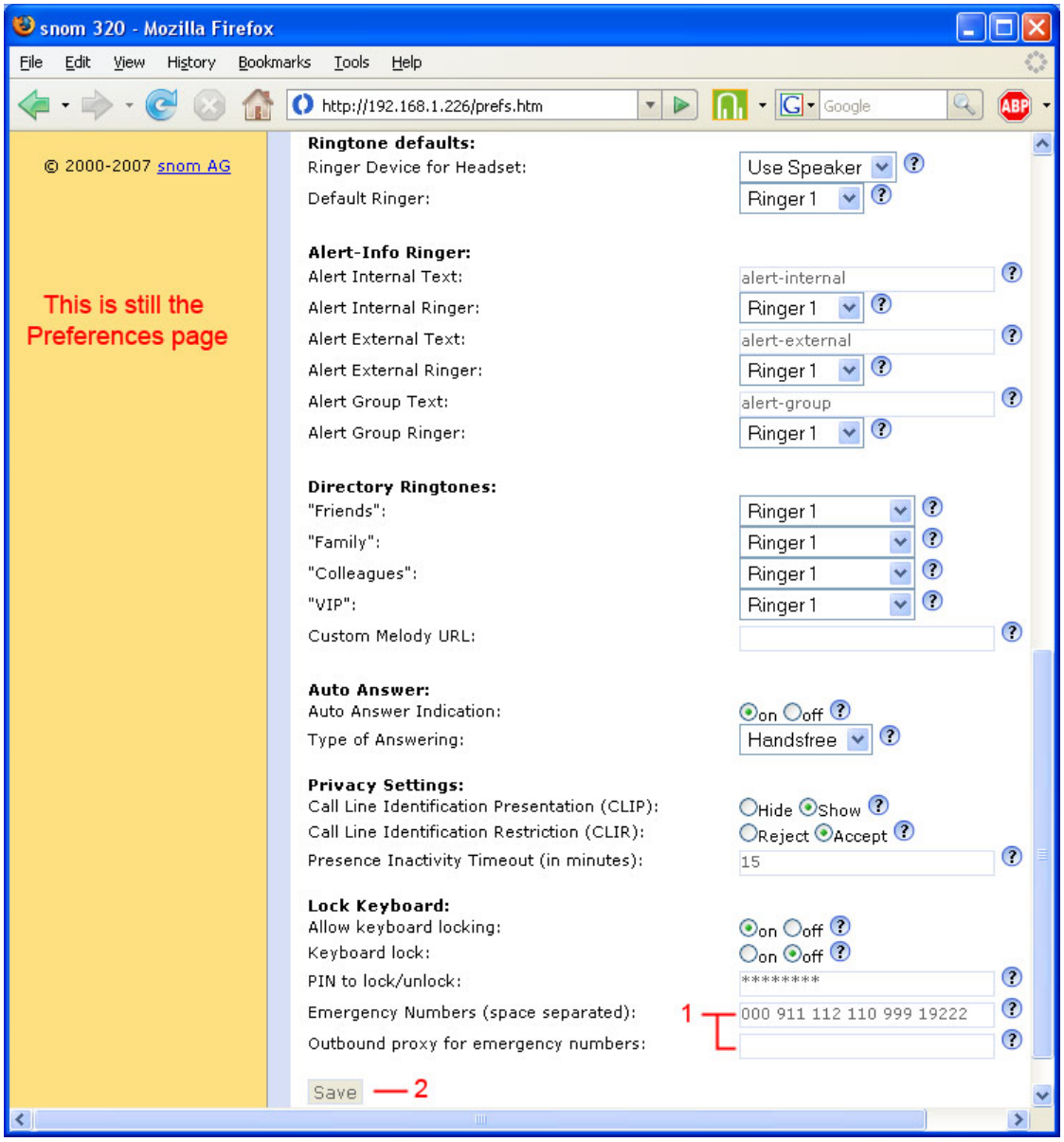

Fig. 4.14 Snom Preferences Menu - 2nd half

This part of the preferences page shows suggested initial settings (most at default). There is a relevant point of note however :

1. **Emergency Numbers** : First the Australian 000 needs to be entered in the list. Then : Emergency services response relies sometimes on known address for action. For the PSTN world this is a given being attached to the CLI of the caller. For the VoIP world this has become a grey area, particularly re portability of SIP devices. Snom phones have a feature where emergency calls can be routed via a different proxy. What this should be is the IP address of an FXO gateway device connected to the PSTN CO port of the service supplying the ADSL/2 feed to which this phone is connected. In this way emergency calls will be forwarded via this port and will bear the PSTN CLI of the relevant premises. The FXO gateway device will need to be appropriately programmed to accept this traffic. From a regulatory standpoint this emergency facility would be considered mandatory. It is currently suggested that a suitable FXO capable device may be the Linksys SP3102 which is detailed following.

2. As before remember to **Save** any settings changes before leaving the page.

Snom 320 Function Keys :

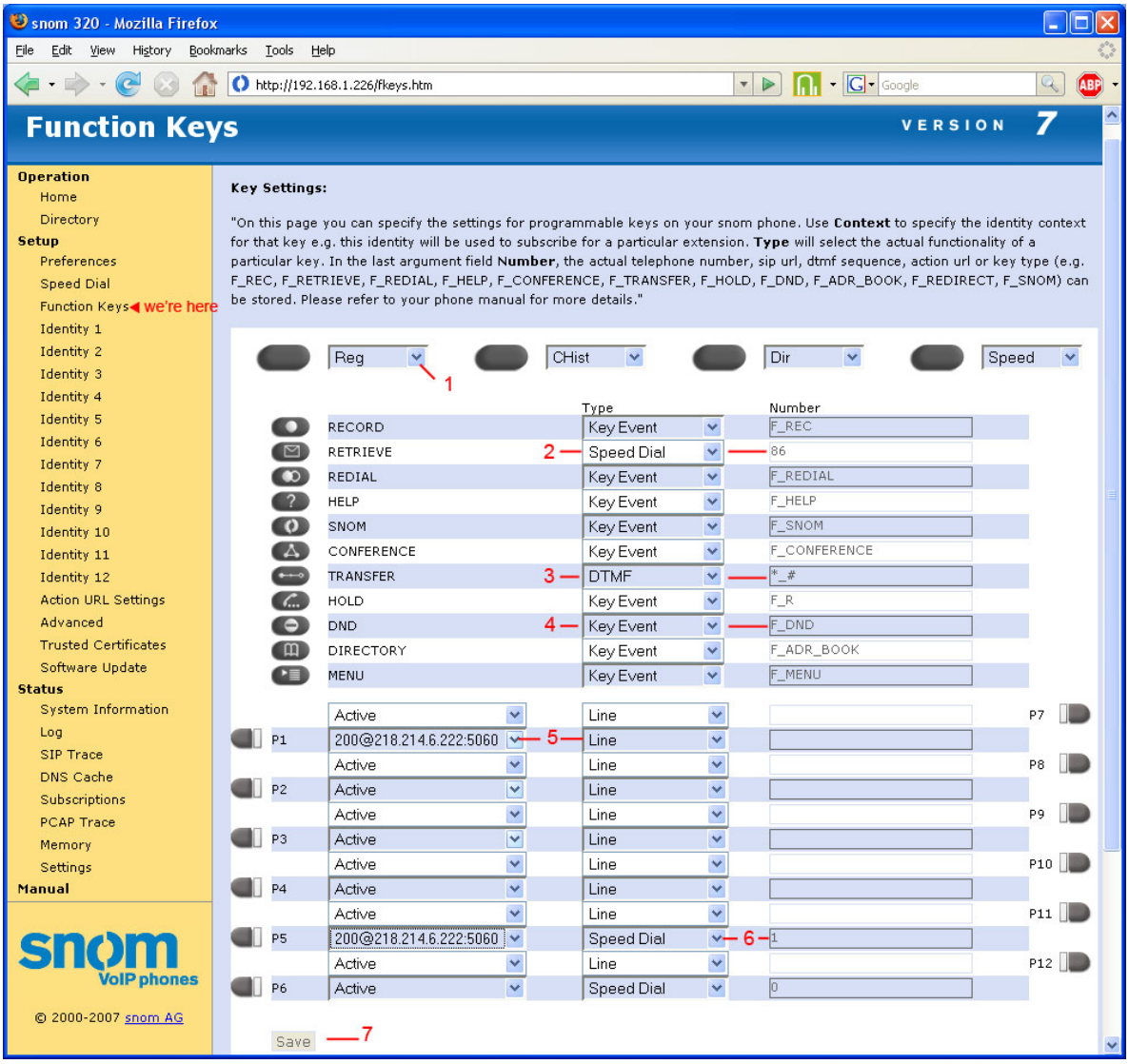

Fig. 4.15 Snom 320 Function Keys

The Snom 320 has a large number of Function keys which are all programmable.
The purpose of this page is to show some re-assignments that are useful to start with :

1. **Soft Key functions** : relative to the Snom display when idle. Can be re-assigned according to the drop menu choices shown when the ellipsis indicated is clicked.
2. **Retrieve** : In this case the key has been reassigned to dial 86 for Hybrex Voice Mail.
3. **Transfer** : The ISU3 function code for Hold/Transfer is *#. So the pre-labeled Transfer key is set to (post) dial DTMF the digits *# to enable this function with the ISU3.
4. **DND** : This key definition is left at default - but the relevant setting is made in Preferences (see fig 4.13 pt 3 pp4-13) to enable this feature. The advantage of using this function this way is the phone displays the DND status as well as the GDS enacting it. The key toggles the state in use, but note the extension state change at the GDS takes a moment or two.
5. **Context** : The unlabeled keys in the lower section can be set to a “context” (using the ellipsis) - this sets the “Identity” to which this key is relative. The example setting shows key P1 to be set to annunciate the “Line” status for the example extension account of this phone (200) on the test ISU3.
6. **Type** : The key “Type” can be set from the drop menu obtained by clicking on the Type field ellipsis for the key. Relevant to the Type a parameter can then be entered in the field to the right. In this case the example shows key P5 set to “Speed Dial” to dial a speed number - 1 in this case - from the settings on the Speed Dial page, and the context is the example identity (200 on the ISU3).
7. Remember to **Save** any settings made before leaving the page.

Snom 300 Function Keys :

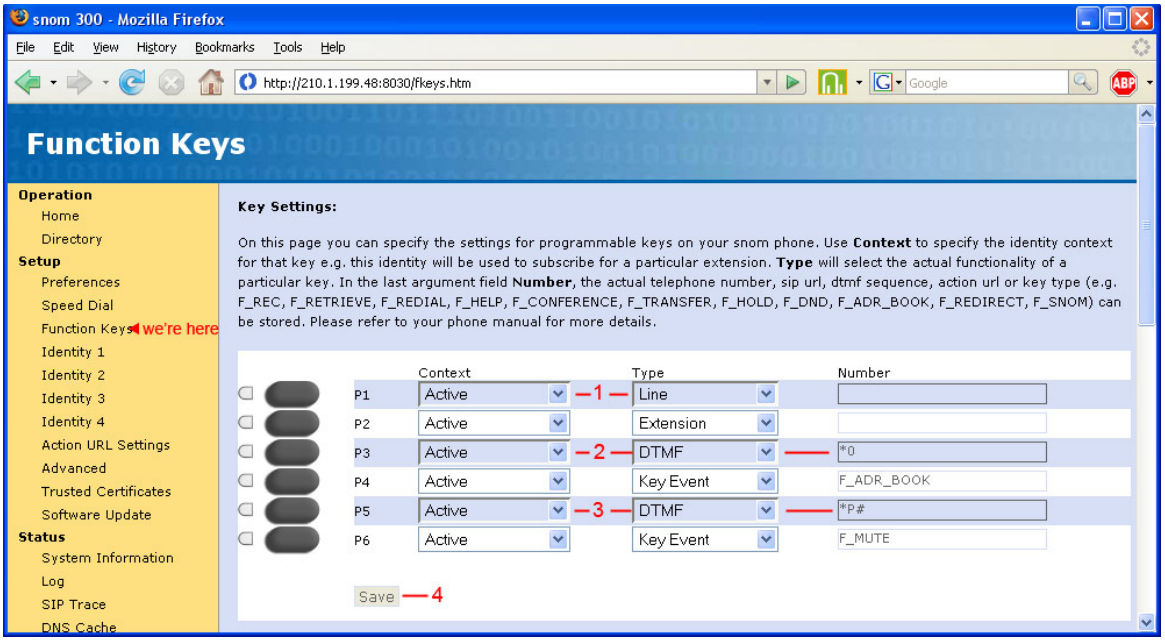

Fig. 4.16 Snom 300 Function Keys

The Snom 300 phone has fewer function keys than the 320; the 300 is of course a simpler and more economical phone. The function keys available however will cater for many needs as they are also programmable and can be put to good use.
The methods noted for the Snom320 also apply here since the phones are of the same software family : ie field values can be set by selection from the ellipsis drop menu where available, or entered directly in the Number field. Context sets the “Identity” (line account) to which the key applies; and Type sets the type of action the key initiates.

1. **Key P1** : is shown set to annunciate the line status of the active account.
2. **Key P3** : is shown in this example to be dialing *0 - the normal GDS “call pickup” code enabling calls ringing on another extension to be picked up.
3. **Key P5** : is shown set to dial *# which is the Hold/Transfer code for the ISU3 - this key would then be labeled “Transfer” on the handset memory card.
4. Reminder to **Save** any settings made before leaving the page.

Snom Firmware Update

For our purpose it is very likely the first step in setting up a Snom phone for use as an IP extension will be to update the phones firmware to the relevant version. At the time of writing the version required will most likely be 7.1.32 or later. The current firmware version of the phone can be checked on the System Information Page :

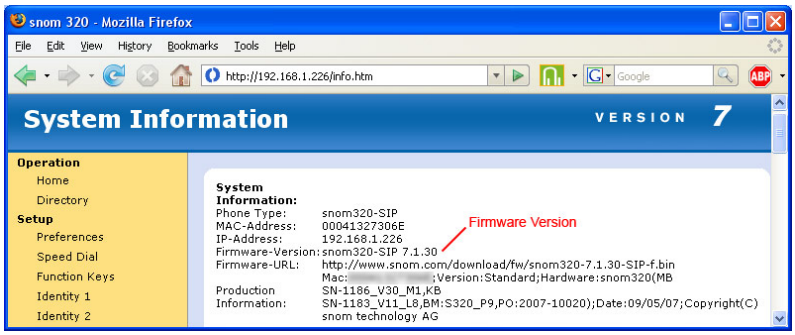

Fig. 4.17 Snom 320 System Info

Updating the firmware is a relatively painless process : Internet access is required. With the phone Web interface showing in one browser window, open another browser window and point it at the Snom Software wizard: http://wiki.snom.com/Firmware/Wizard.

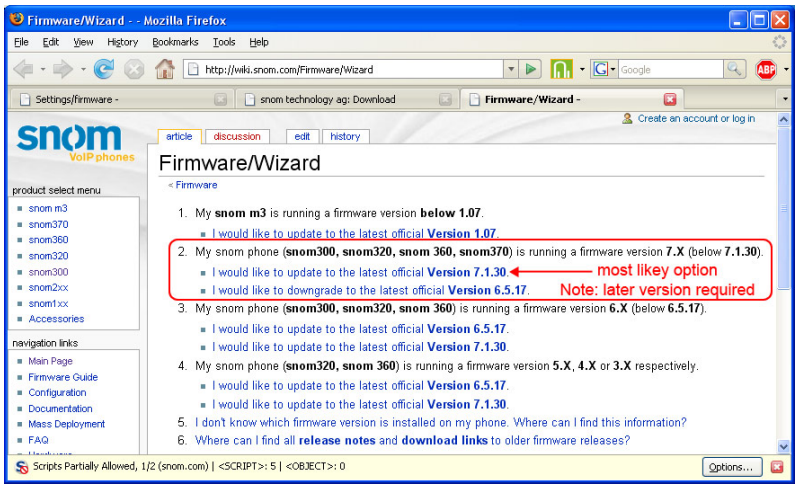

Fig. 4.18 Snom System Firmware Update

Click on the relevant link and you will be taken to the download page for the firmware. In this example it is the page http://wiki.snom.com/Firmware/V7/Update.
This page contains the instructions for updating your phone.

Choose the relevant link for your phone, right click it and select “Copy” from the context menu.

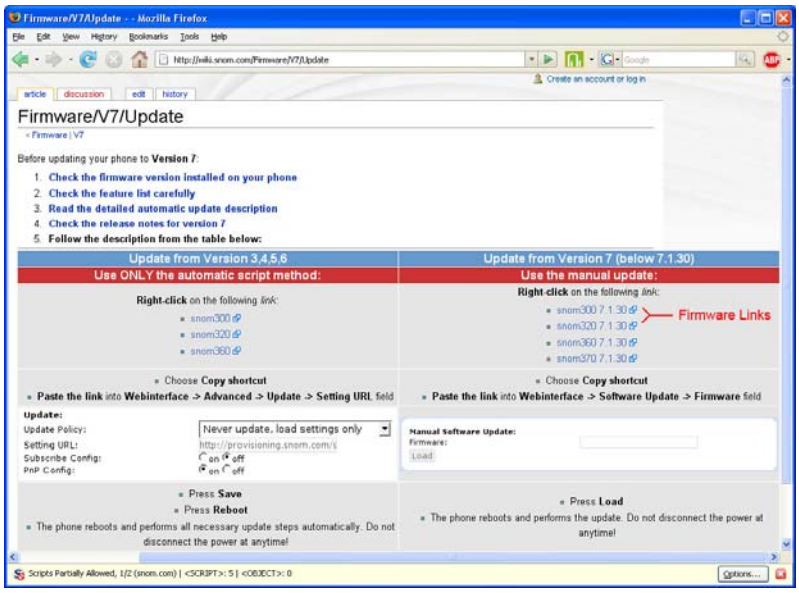

Fig. 4.19 Snom System Firmware v7 Update Page

Then right click and select “Paste” from the context menu. This will paste the correct link that you have just gathered for the firmware in place. Check to make sure you didn’t miss any characters when you selected.

Then click the Load button to start the update.
It is VERY important that you DO NOT Remove power from the phone now until the update is complete and the phone re-booted. Otherwise you will very likely end up with a brick instead of a phone.

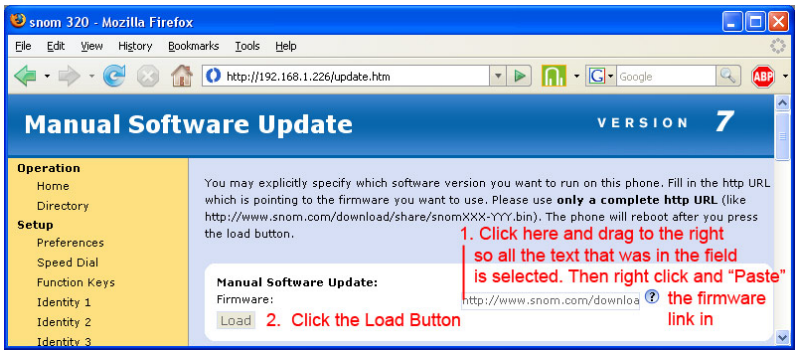

Fig. 4.20 Snom Phone Firmware Update Page

Snom Notes

DTMF Issue :

At the time of writing the latest available firmware version for the Snom300 and the Snom320 phones is ver. 7.1.30. This version has one known issue with post dial DTMF in that when the phone is programmed for SIP Info style DTMF, as we need here, the DTMF is sent as both Inband (as tones with the voice) and SIP Info messaging. The ramification of this is that processing at the receiving end sees two separate tone bursts generated for each user button press. So when presented to an IVR or DISA interface the expected functionality is lost.

This has been reported to Snom and modifications have been made to the firmware to correct this issue and are currently under test/QA. The expected firmware version for this correction is 7.1.33 which we are waiting for Snom to release. Hence the previous page instructions will most likely be required for the upgrade when it is available.

{Stop Press: A beta firmware version has been trialed here that fixes this issue 30Mar08}

Snom Versatility :

As can be deduced from the previous pages the Snom firmware allows a great deal of versatility. For example the Snom300 has 4 "lines" - these are conceptually classed as "identities". With one line registered to your main GDS ISU3, a second line can be registered with an ISU3 at a second business site, a third line could be registered with a VSP (OurIpTel, Exetel, Sylantro, FWD, etc) for low tariff personal calls, and the fourth line registered with your on-premises FXO gateway for emergency calls. Calls can be taken on any identity that is registered. Dialing out occurs on the "line" or "identity" you have set as active (with the exception of the mentioned emergency calls if that option is configured) and setting an identity as active takes two or three key presses on the phone. The Snom300 has six DSS keys (w/- LEDs) which are completely programmable - so you can have line appearances and/or set up many functions as "one touch" buttons. The Snom320 has twelve useable identities, 12 blank DSS keys (w/- LEDs) and another 11 (or 15 counting softkeys) keys which are labeled - all are programmable.

Dialing Numbers :

There are a couple of points that need to be re-mentioned regarding dialing of numbers. The first point is that the Snom phones have a “tick” key - this has a number of uses - as a confirmation key for menu choices for example. However when dialing it is regarded as the “send” key. If the dial timeout (Auto Dial) is set on the Advanced Settings > Behavior tab the number will be sent when this time has elapsed since the last key-press. In any case when the “tick” key is pressed any number dialed is immediately sent.

The second point is when any PSTN number is dialed a trunk on the GDS must have been selected. There are many ways to do this but the most common way is the “dial 9” method where the digit 9 is pre-pended to the dial string therefore pre-selecting a trunk from the preferred group. For incoming calls with CLI (see also below) it is this 9 pre-pend that prevents dialing numbers direct from Missed and Received Call records of the Snom phone (if they are enabled). Dialing from these records requires noting the number and pre-pending the 9 before dialing out. Of course the Dialed Numbers records will show numbers with the 9 pre-pend and these can be dialed from directly.

CLI with ISU3

At the time of writing there is an issue with GDS ISU3 CLI. The issue is that callers CLI is only properly recorded by the Snom when the parent GDS is set to show number only and the incoming call only bears a CLI “number”, and the call is a DID or the Snom extension no. is ring assigned to the incoming GDS trunk. Where GDS “Name” or “Name and Number” is enabled and the callers number has a name match in system Speed Dials, or the call incoming is a mobile call on a Telstra PSTN trunk (where the name “Mobile” precedes the callers number) then what is received as CLI is only the first word in the name string whatever the source. This is a protocol issue that is currently under review with a solution hopefully forthcoming.

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Analogue Telephone Adapters (ATA)

The main type of SIP endpoint device preferred for use with our ISU3 will not be an Analogue Telephone Adapter due to obvious usage needs and the advantages of using a more compliant SIP IP phone, not to mention the interoperability problems that will be encountered. There is however one device that will be detailed for special reasons. One because it works fairly well as an extension device even though there is still a niggle in its operation. But more importantly, two, because it can provide an FXO gateway for the previously mentioned Emergency Services feature of the Snom phones. Re-stating this feature : the Snom phones have a provision for definition of emergency services numbers that can be dialed even when the keypad is locked. Alongside this feature is the ability to specify an alternate proxy IP for emergency dialing to be sent to. This alternate proxy is preferably an on-premises FXO gateway connected to a PSTN line. In this case the FXO gateway is the device described below. Thus the Emergency Services have the correct CLI for the call in the event that an Ambulance, or whoever, is needed and can respond appropriately.

Linksys SPA3102 ATA

The Linksys SPA3102 is a very versatile device for an ATA. It is extremely programmable as you will discover when faced with its Web console in Admin-Advanced mode. For our uses though it has an FXO port, an FXS port, two VoIP ports, and fundamental router features that enable it to be set up as an IP extension endpoint, and/or an on-premises FXO gateway for emergency or other uses. There are many other features such as T38 fax support, PSTN failover, PSTN to VoIP gateway, etc, that can be set up.

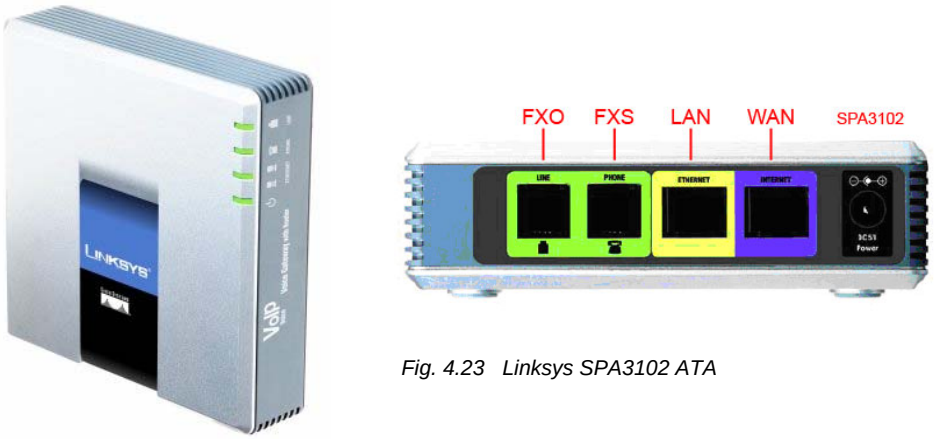

Fig. 4.23 Linksys SPA3102 ATA

For our purposes the SPA3102 will be set up as a LAN device behind a modem router (eg Draytek) as that is all that is necessary for the required functionality. Note that this will be extension site equipment primarily, and although this does not preclude installation on a network carrying an ISU3 such a placement would not normally be the case.

So to set up: (In the following : the term SPA will be taken to mean the SPA3102).

- Connect the SPA WAN port to your LAN switch or router using a standard Ethernet patch cable and power it up.
- Connect an analogue handset (SLT) to the SPA Phone (FXS) port.
- Lift the handpiece of your SLT and dial **** (four stars) to get the SPA's IVR voice prompt menu. The voice will prompt you to dial a code for the desired function. These codes are detailed in the small booklet that comes with the SPA if you are interested.
- Dial 110# (not too quickly) and listen - the voice will tell you the WAN IP the SPA has acquired via your routers DHCP function : make a note this IP as this is what you will use to program the device. This is presuming of course the router has DHCP serer enabled - if not it needs to be so.
- Programming the SPA is detailed following :

SPA3102 WAN Port IP Setup

- Connect your PC to the LAN the SPA is now connected to.
  If your PC has a compliant IP address then OK, otherwise the easiest is to set it to "Obtain an IP address automatically" for now.
- On your PC launch a browser and browse to http://[WAN IP of the SPA]
  The WAN IP you will have obtained via the SPA's IVR voice menu (see previous page).
- At default the SPA has no password set so you will be presented with the SPA Web console immediately. To do the necessary programming you will need to set the SPA into **Admin** and then **Advanced** mode - you will find links near the top right corner of the Web console to click on to do this.
- The SPA default IP address method for the WAN port is DHCP. This is fine for initial setup but if you want to be able to remotely access the SPA later on, by port forward through the network's router ('Net gateway). Then you will need to fix the WAN port IP address. To do this click on the **Router** tab, then on the **Wan Setup** tab. : -

Fig. 4.24 SPA3102 Web Console : Router >Wan Setup page

If you wish to fix the WAN IP of the SPA:

1. Set Connection Type to Static
2. Enter desired IP Address, Netmask, and Gateway.
   Note: check the address assigned on the LAN Setup page first - the SPA will get into a knot if you set both LAN and WAN ports into the same subnet.
3. Submit changes - the SPA will reboot, and you will need to re-set the address in your browser to the new address to log back onto the Web console.

Notice the SPA can also be set to use PPPoE login from behind a bridge mode modem.
The other tabs show the basic WAN to LAN router functions - noted out of interest.

SPA3102 Setup for IP Extension Use

A. : Voice - SIP tab :

Continuing from the previous pages and presuming your PC is connected to the SPA via the network and you are viewing the SPA's Web console in Admin + Advanced mode:

- Click on the Voice Tab - then click on the SIP tab :

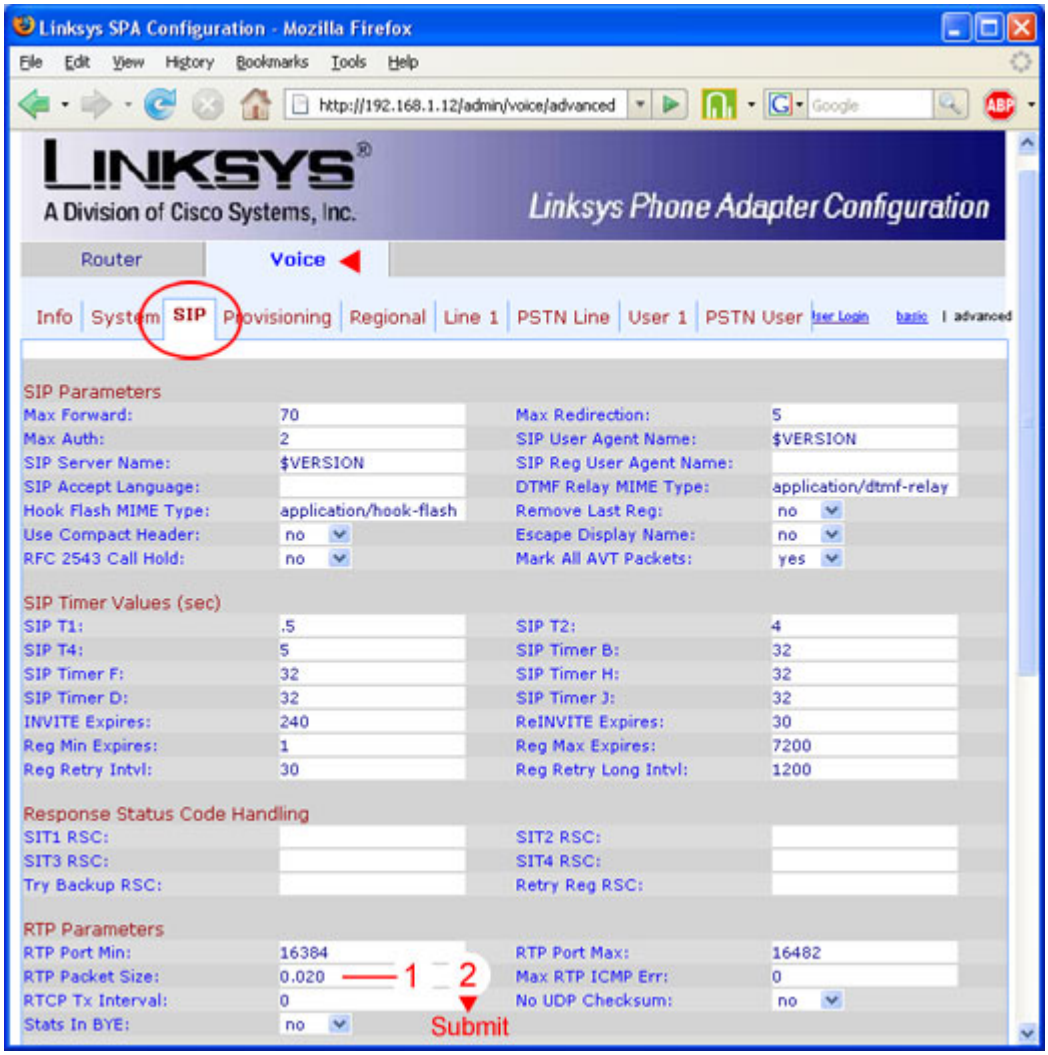

Fig. 4.25 SPA3102 Web Console : Voice > SIP setup page

1. **RTP Packet Size** : This is the amount of voice sent in each packet.
   Like the "Voice Frame per Packet" setting on the ISU3 DSP Settings page.
   The field unit is seconds. So the value shown is 20mS which is the recommended.
   Other values can be set eg 0.030 (30mS), 0.040 (40mS) to reduce bandwidth but as noted previously any variant must be tested.
   Special note here : this value controls RTP on all interfaces whatever the codec selected. It is known that Snom phones will not operate with G729 @ 30mS so if you are setting up this gateway for the emergency FXO proxy use and intend to use G729 for the codec for this SPA then you must set the above to 0.020 (20mS) as recommended or the Snoms will not transmit voice.
2. Remember to **Submit** before leaving the page - SPA will reboot but you will have the Web console back in 5 seconds or so.

   Ancillary note: Out of interest you will notice at the very bottom of this page there are controls that allow you to set up the SPA for WAN IP sharing in the same way that the ISU3 can be set up (type 2 - NAT'd).

SPA3102 Setup for IP Extension Use . cont..:

B. : Voice - Regional tab :

- Click on the (Voice) Regional tab: (Web console is still in Admin - Advanced mode)

Fig. 4.26 SPA3102 Web Console : Voice > Regional setup page

1. **FXS Port Impedance** : For Australia the regional impedance is 220 + 820||120nF. Set this to give the best impedance match to Australian Telephone handsets.
2. **FXS Port Gains** : Both Input and Output gains can be set here if desired. If the phone voice levels at the handset attached are not acceptable they can be adjusted here. Default shown.
3. **DTMF Tone Level and Length**: can also be adjusted here if required. Default shown.
4. **Submit** changes - the SPA will reboot and you will have the Web console back in 5 seconds or so.

It will be noted that there are a great number of factors that can be adjusted on this page. Of interest may be the call progress tones which can be adjusted if required. There are also service codes which apply to features built into the SPA which we will not be using for now. You may refer to the SPA3102 user manual if you want to alter any of these factors. For now these can be left at default as we will not be using them for this implementation.

SPA3102 Setup for IP Extension Use . cont..:

C.: Voice : Line 1 tab : (Part 1)

- Click on the (Voice) Line 1 tab: (Web console is still in Admin - Advanced mode)

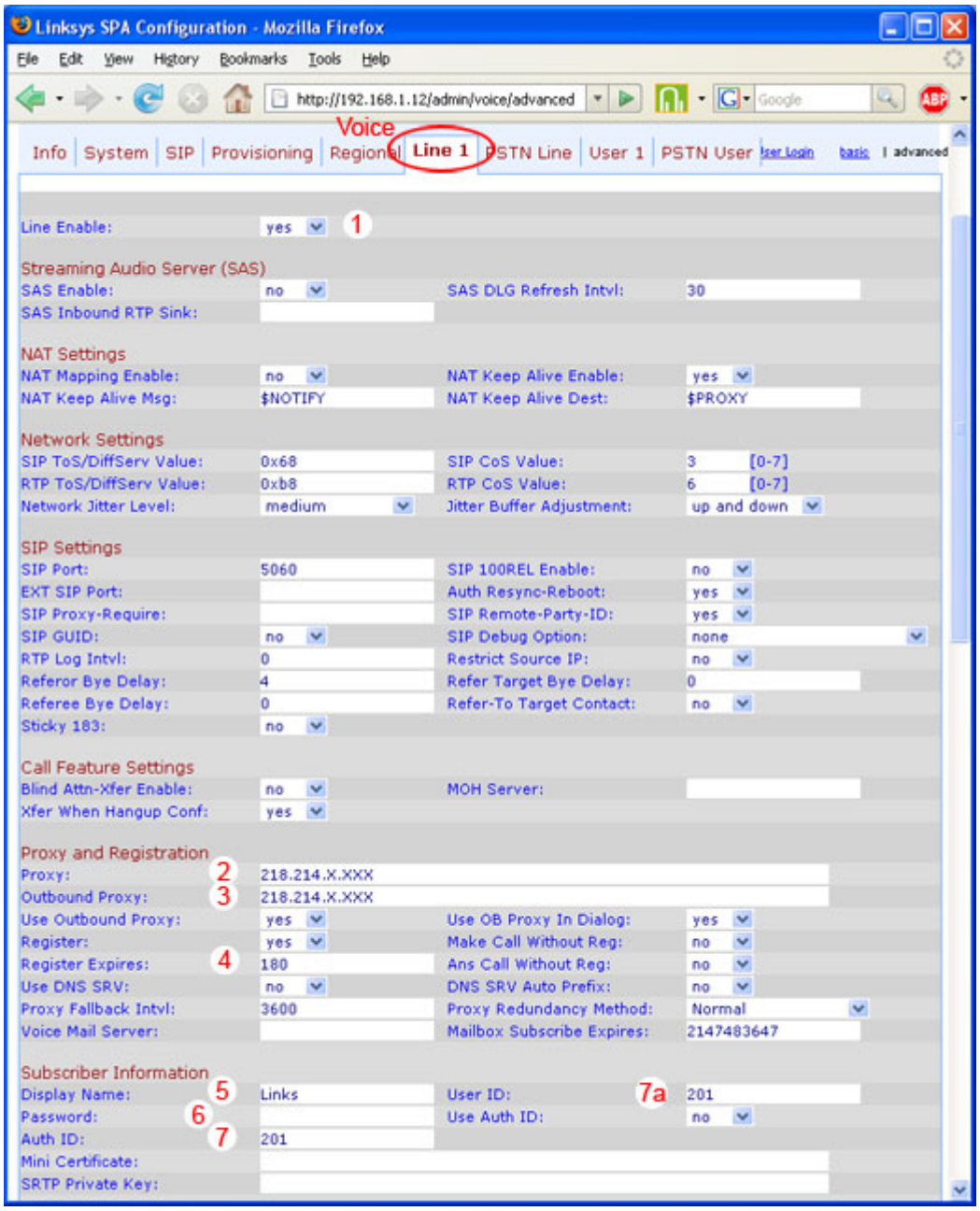

Fig. 4.27 SPA3102 Web Console : Voice > Line 1 setup page (part 1)

1 **Line Enable**: = Yes
2. **Proxy** : Is the SIP Registrar WAN IP - Enter the WAN IP in use for the ISU3 card.
3. **Outbound Proxy** : Is where IP traffic will be sent - Usually this is also set to be the ISU3 card WAN IP.
4. **Register Expires** : The registration interval - if this SPA is situated behind a router (NAT'd) this value will be set to less than the routers port binding time so that the ports are kept open for incoming traffic. Less than 3 minutes (180 sec) is generally OK
5. **Display Name** : For reference - is usually sent in SIP Invite as additional caller ID info.
6. **Password** : For the ISU3 there isn't one.
7. **& 7a. Auth ID and User ID**. These are set to the GDS/ISU3 extension number this device is registering as.

SPA3102 Setup for IP Extension Use . cont..:

C.: Voice : Line 1 tab : (Part 2)

- Continuing from the previous page on the Voice > Line 1 page (lower half) :

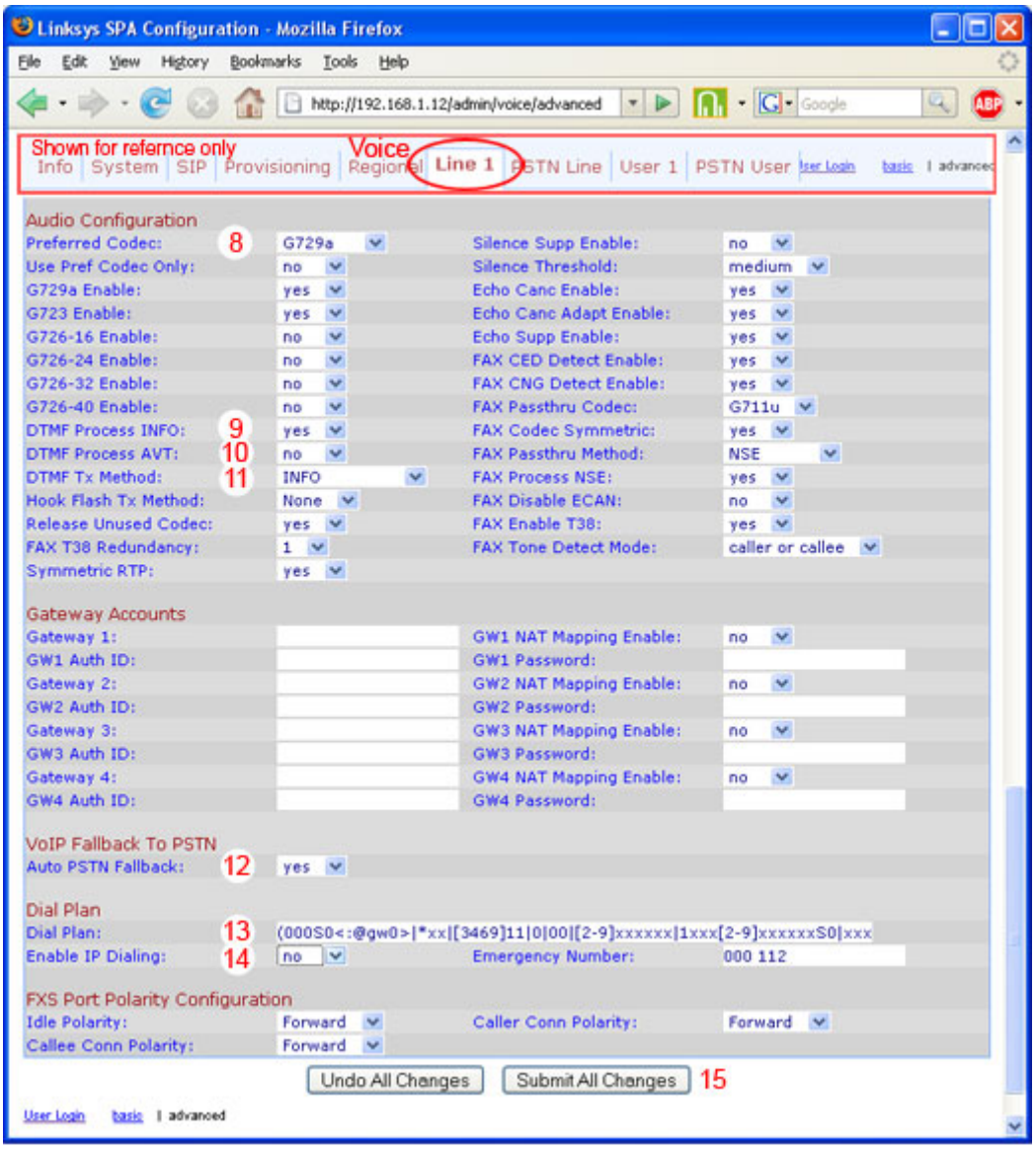

Fig. 4.28 SPA3102 Web Console : Voice > Line 1 setup page (part 2)

8. **Preferred Codec** : Set as desired. G729 is recommended. See Appendices : Codecs
9. **DTMF Process Info** : Set to Yes to process incoming SIP Info DTMF (nb:untested)
10. **DTMF Process AVT** : This is Linksys' way of saying RFC2833 DTMF. Set to No.
11. **DTMF Tx Method** : Post Dial DTMF sending - Set to INFO for ISU3 post dial and *# transfer function to work.
12. **Auto PSTN Fallback** : Set to Yes if you have connected SPA Line (FXO) port to PSTN CO. (trunk) Then when the SPA cannot register, or Internet service is unavailable, phone plugged into SPA Phone (FXS) port will automatically be connected to PSTN.
13. **Dial Plan** : Specifies what dialing is accepted. Add the first term "000S0<:@gw0>" to send emergency calls out the Line (FXO) port - Emergency Services measure - like 12 above, the FXO must be connected to PSTN for this to work. Other emergency numbers can be added in a similar fashion. The rest of the Dial Plan terms shown are the SPA default set. See Appendices : Dial Plans.
14. **Enable IP dialing** : If you have any speed dials set up that dial a strictly IP Number, like "sip:john@sipserver.com" etc. then this must be set to Yes for them to work. Otherwise set to No as the function isn't needed.
15. **Submit** : Click to submit changes - SPA will reboot and be back in 5 seconds or so.

Notes on Using SPA3102 as an IP Extension device :

The SPA3102 is a very facile little device as you would have gathered seeing the number of configuration items in its Web console. It can be programmed to do a great many more things than what is shown in these pages. There are a few things about it's operation that I have noticed so as a guide the following deserve a mention:

Dialing :
For the use in question an analogue (SLT) phone is connected to the SPA. When dialing the SLT produces DTMF tones. The SPA is however a SIP device that sends messages in SIP protocol, it is also for our uses programmed to send post dial DTMF in SIP Info protocol. To do this the SPA first detects DTMF tones before converting to the necessary numbers, it is also interesting that in the post dial function the DTMF tones from the phone are blocked by the DSP in the SPA so two tones for every key press are not received at the other end.
The effect of all these smarts is that if numbers are dialed too quickly they will often be missed by the SPA. So **dialing needs to be slow and deliberate to be successful**. In the post dial sense I have noticed a small part of the first or second digit will reach the other end and depending on the efficiency of the DTMF detectors there these may be detected. So the current recommendation I would have is that using this device for Internet Banking for example may not be a really good idea.

Dial Plans:
The SPA has four interfaces including the FXS and FXO ports. The placement of calls to any of these interfaces can be managed using the Dial Plans - in this case the Line 1 Dial Plan. There may be a workaround here to the above Internet Banking example in that the SPA can be set up to be able to access the connected PSTN line by dialing a prefix for such uses. In fact the main reason for detailing this device is its use as an FXO gateway for emergency services - for which the effect of allowing other uses like that above would have to be considered. In any case there are notes in the appendices on Dial Plans should you wish to pursue these.

Service Tones :
You will have noted on the "Regional Settings" page there are a number of tone settings that can be configured. A temptation arising from this is to set the service tones to be equivalent to Australian tones which is easily done. A brief note I would make about this is :the SPA doesn't indicate visually when Internet Access or Registration is lost, and in these cases it will automatically connect its FXS (Phone) port to its FXO (Line) port - a "lifeline" function.
If the SPA's dial tone is set to the same as Australian dial tone then there is often much confusion as to why dialing extension numbers say in our case don't work any more. If the SPA's dial tone is different to Aust dial tone the loss of 'Net service as above is immediately apparent so confusion is averted.

Supplementary Service Codes :
As part of the functionality of the SPA there are a number of service codes implemented - the dialing of *69 for call return for example. These codes are defined in "Regional" settings and controlled in the "Line 1" and "PSTN Line" pages. These have been primarily left at default in this document as Hybrex doesn't generally use * codes. If however there is a conflict these codes can be changed or disabled on those pages.

Codec Use :
One of the points about codec use to be aware of is that the SPA has only one codec license for G729. In addition to this if G729 is defined in any interface codec preferences, even if it isn't actively used, it is in effect consumed any time that interface is in use. This is a little unfortunate as G729 is the preferred codec for many uses. Another effect of this is when G729 is consumed in this way there cannot be two instances of either G723 or G726 (which we don't use) in operation simultaneously. G711a and G711u don't suffer any constrictions in this way. This is one of the reasons the FXO Emergency Services gateway implementation described following is recommended to use G711 codec - the other is voice quality : considering the LAN only traffic path G711 is better for this use.

The SPA3102 as an Emergency Services FXO Gateway - Setup:

Continuing from the previous section - presuming SPA Web console is being accessed in Admin > Advanced mode - (refer to pp 4-20, 4-21 for initial setups if necessary).

PSTN Line Settings (part 1) :

- Click on the Voice > PSTN Line tab :

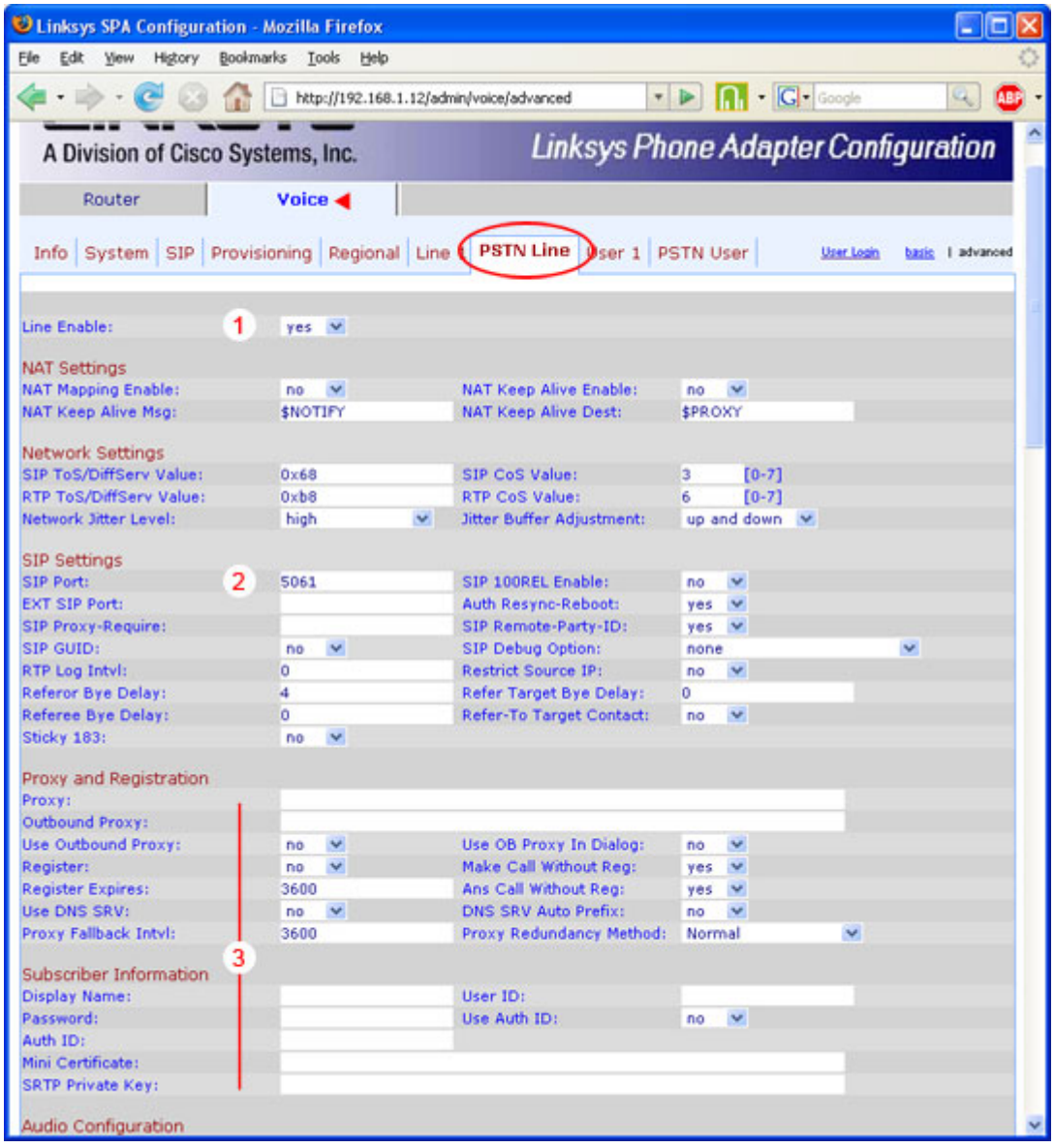

Fig. 4.29 SPA3102 Web Console : Voice > PSTN Line setup page (Part 1)

The PSTN Line tab refers to the SPA FXO port functions and settings:
For all these to mean anything the FXO (Line) port must be connected to a PSTN CO feed.
Suggestion is for this use is this feed be the PSTN service carrying the ADSL connection.

1. **Line Enable** : Set to Yes
2. **SIP Port** : Note SIP port for access to the FXO gateway is 5061 (2nd SPA VoIP port)
3. **Proxy / Registrar / Subscriber** : Noted here as a matter of interest - this FXO port can be registered with SIP server/ SIP provider/IPPBX for use as a gateway.

Rest of the PSTN Line page continued over ..

SPA3102 Setup for FXO Emergency Gateway Use . cont..:

PSTN Line Settings (part 2) :

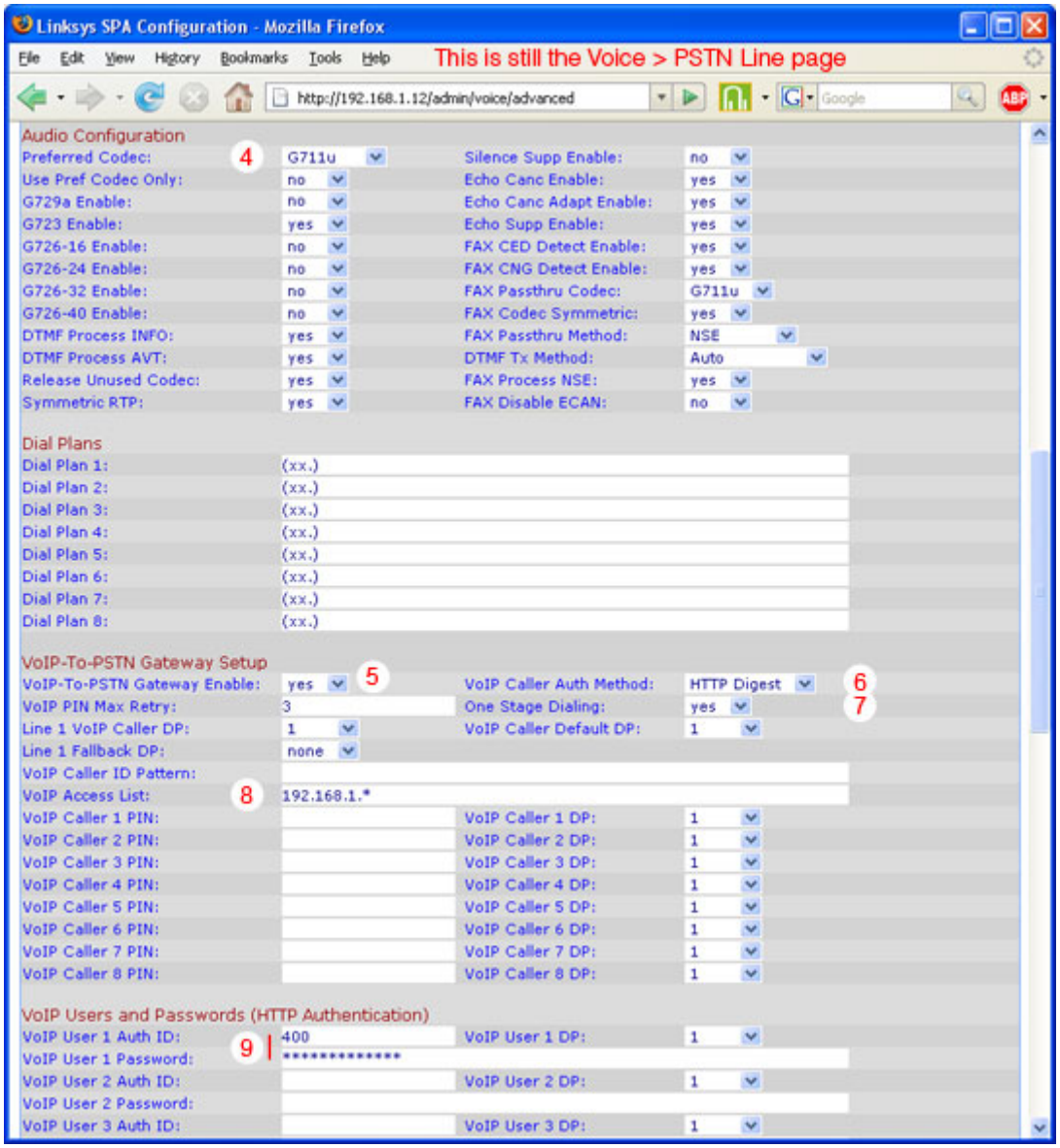

Fig. 4.30 SPA3102 Web Console : Voice > PSTN Line setup page (Part 2)

4. **Preferred Codec** : Set to G711u - LAN use for local ES gateway function can withstand bandwidth requirement and voice quality is best.
5. **VoIP to PSTN Gateway Enable** : Set to Yes - for purpose of this implementation.
6. **VoIP Caller Auth Method** : Set to HTTP digest so only authorised VoIP users can hop off to PSTN from here. (main line of defense against fraudulent use).
7. **One Stage Dialing** : Set to Yes - necessary for 000 calls to work properly.
8. **VoIP Access List** : VoIP calls originating from IP's in this (comma separated) list will bypass any authentication requirement for making calls out PSTN port. Shown is example : allowance for devices situated on a local LAN subnet 192.168.1.xxx.
9. **VoIP Users and Passwords** : Authenticated users list - username/passwords - for use with VoIP Caller Auth Method above. Shown is example of username 400 and pwd.
   This may be another VoIP user on a network not covered in the Access List. Note this is not a registration - the user is challenged for authorization when they ask to make a call (via SIP Invite) out the PSTN port.

Continued over ...

SPA3102 Setup for FXO Emergency Gateway Use . cont..:

PSTN Line Settings (part 3) :

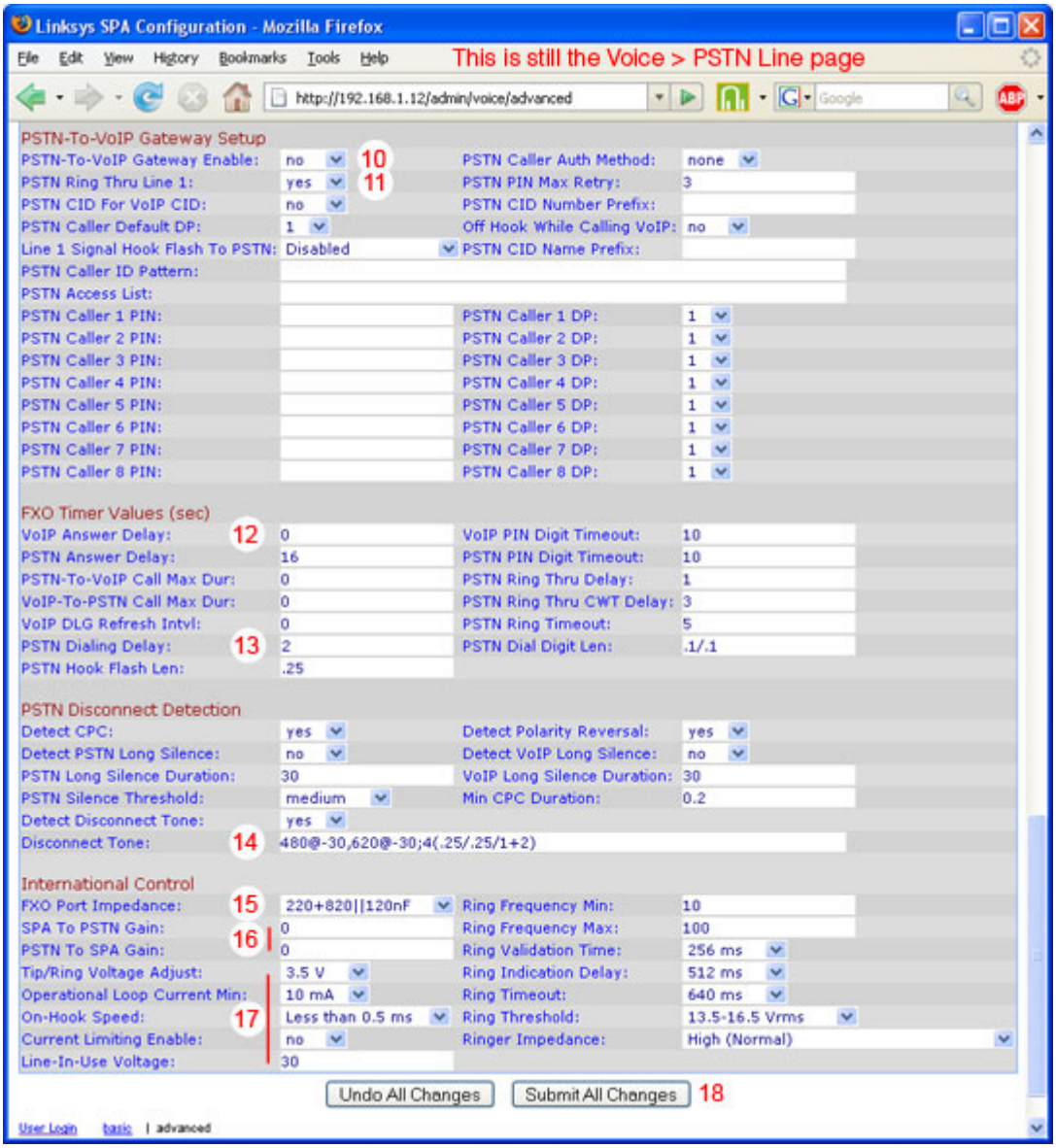

Fig. 4.31 SPA3102 Web Console : Voice > PSTN Line setup page (Part 3)

10. **PSTN to VoIP Gateway Enable** : Currently set to disable as we are not using.
11. **PSTN Ring Thru Line 1** : Set to Yes to enable incoming calls to go to attached SLT handset (if it exists which it should in practicality).
12. **VoIP Answer Delay** : Set to 0 - VoIP calls to FXO will pick up immediately after Auth.
13. **PSTN Dialing Delay** : Set to 2 (seconds) - good value to allow exchange readiness for dialing.
14. **Disconnect Tone** : If required - disconnect tone settings can be made here for PSTN port clear down.
15. **FXO Port Impedance** : Australian PSTN port impedance is 220 + 820||120nF - set.
16. **Gains** : Both incoming and outgoing line gains can be set here to adjust audio levels.
17. **Other Regional FXO Interface controls** : The SPA can be set for port loop current, loop voltage, and hook speed. In addition the parameters for judging when another user is using the PSTN line can be set. Also the ring parameters can be adjusted. Shown are the defaults which appear to work ok.
18. **Submit** : Click submit to save settings - SPA will reboot but be back online in about 5 seconds.

This completes the basic setup of the SPA3102 for Emergency Services local FXO port.
Password Note: One last point : go to the System page and set an admin password and note it !!! … The detail of Snom phone settings to utlilise this FXO function.

Snom Phone Settings for: SPA3102 as a Local Emergency FXO Port :

The details relative to Emergency Services calls for Snom Phones are set on the Snom Web console "Preferences" page : (for more information re the Snom Web console see the Snom section of this manual)

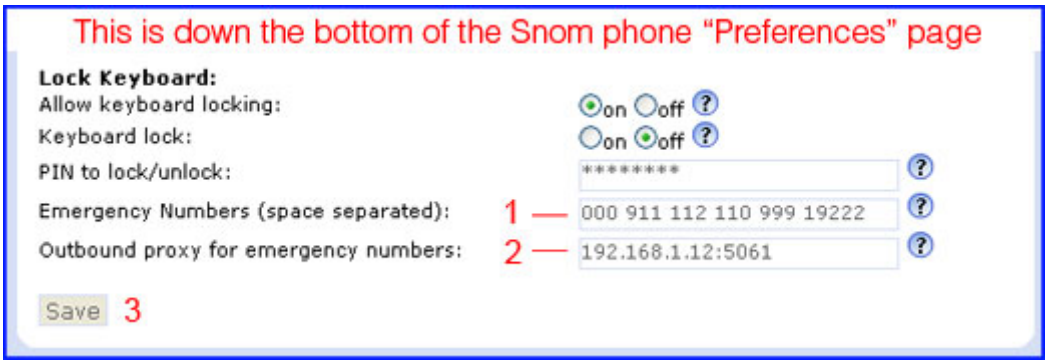

Fig. 4.32 Snom 320 Web Console : Preferences page - excerpt

1. **Emergency Numbers** : Enter here the numbers that will be used to dial Emergency Services. In this case the number 000 is added for Australia. The rest are there at default but are left in as they may be used. For example due to the influence of U.S. television some people here think 911 is our emergency number.
   These numbers can be dialed and the call sent even if the Snom keypad is locked.

2. **Outbound Proxy** : This will be set to the IP address of the SPA3102 WAN Port which will logically be located on the same LAN as the Snom Phone - example only shown. Notice the port number 5061 for the SPA's FXO VoIP interface - this must be used.

3. **Save** any settings changes before leaving the page.

Notes:

Another setup that could be programmed is to set one of the "Identities" on the Snom phone with the details of an/the SPA3102 - IP address:port, and particularly the Username/password of a VoIP User (HTTP Auth) setting - as detailed point 9 on SPA PSTN Line page pt 2 shown previously. This would allow calls to be made to PSTN via the SPA FXO port as required. To make calls in this way the Snom Identity so set (as above) needs to be "activated" first - this is a couple of key-presses on the Snom320, unfortunately on the Snom300 the Web console has to be accessed to make this change. Activation in this sense means calls will be made via the Proxy that is specified in the "active" Identity.
For an SPA on the local LAN this setup is easy. If it is envisaged that calls will be made via an SPA3102 located in a remote network, then that SPA WAN port must be accessible. In other words it must have a public IP - either by assignment or PPPoE - or it must be set up for IP sharing as mentioned previously. This particular setup is not documented here but mentioned as a point of interest.

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Softphone

At the risk of over stating the obvious : a softphone is a piece of software that is installed on your Desktop PC, Notebook PC, and in some cases WiFi enabled PDA, that enables you to make SIP telephone calls from the PC/PDA using an attached headset or other device for the audio. In some cases (PDA) this might be known as a “SIP Client” as it provides SIP protocol connections necessary to make VoIP calls and will use WiFi or other wireless media for the connection.

The SJphone Softphone.

A compatible softphone for the ISU3 is the SJphone softphone as it has the ability to dial digits during a call, and is SIP Info DTMF capable, so the transfer function can be enabled by dialing *# .
The version known as compatible is ver 1.65 which can be downloaded from http://www.sjphone.org/sjp.html. This version is for Windows XP sp2. Compatibility of other versions is unknown at this time.

SJphone Installation

Having downloaded the SJphone install exe from the link above (or other) the initial install process is fairly simple.
Double click on the install exe and you will be issued with a number of dialogs to click [Next] to, with the only exception being to agree to the EULA (End User License Agreement):

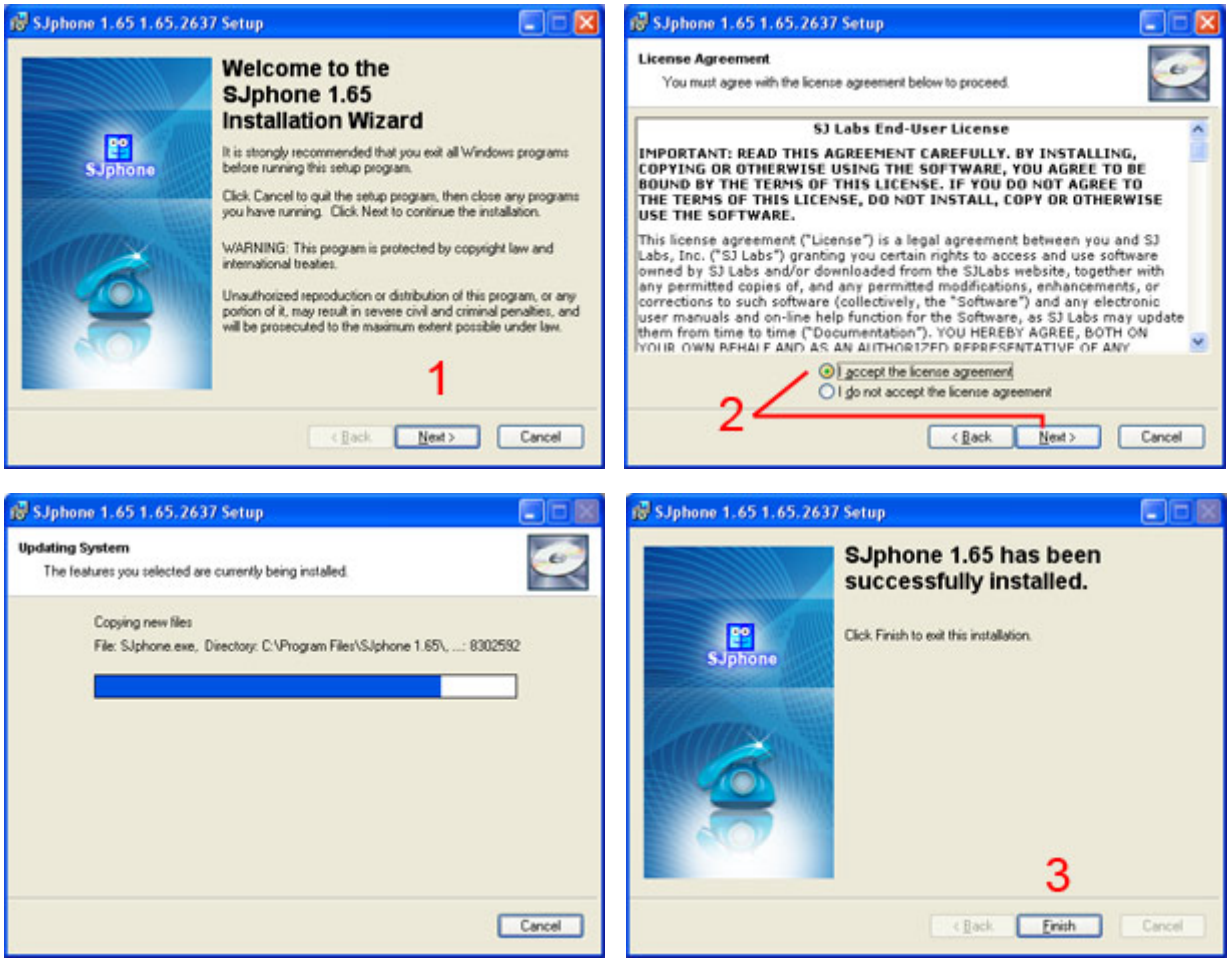


Fig. 4.35 SJphone Install Sequence Screens

SJphone Audio Wizard :

In the next stage of the setup process you will be presented with the Audio Wizard to work through to set up your system audio for the softphone. Note since the common use will be notebook PC which have only one set of audio hardware the softphone will subsume this audio hardware and any audio devices connected.
The Audio Wizard constitutes 7 steps which are explained on the dialogs. Be aware though that you will likely have to dive into system settings along the way to get a good result.
Don't fret too much about getting everything right at this stage as the Audio Wizard can be run again at any time from the softphone interface.

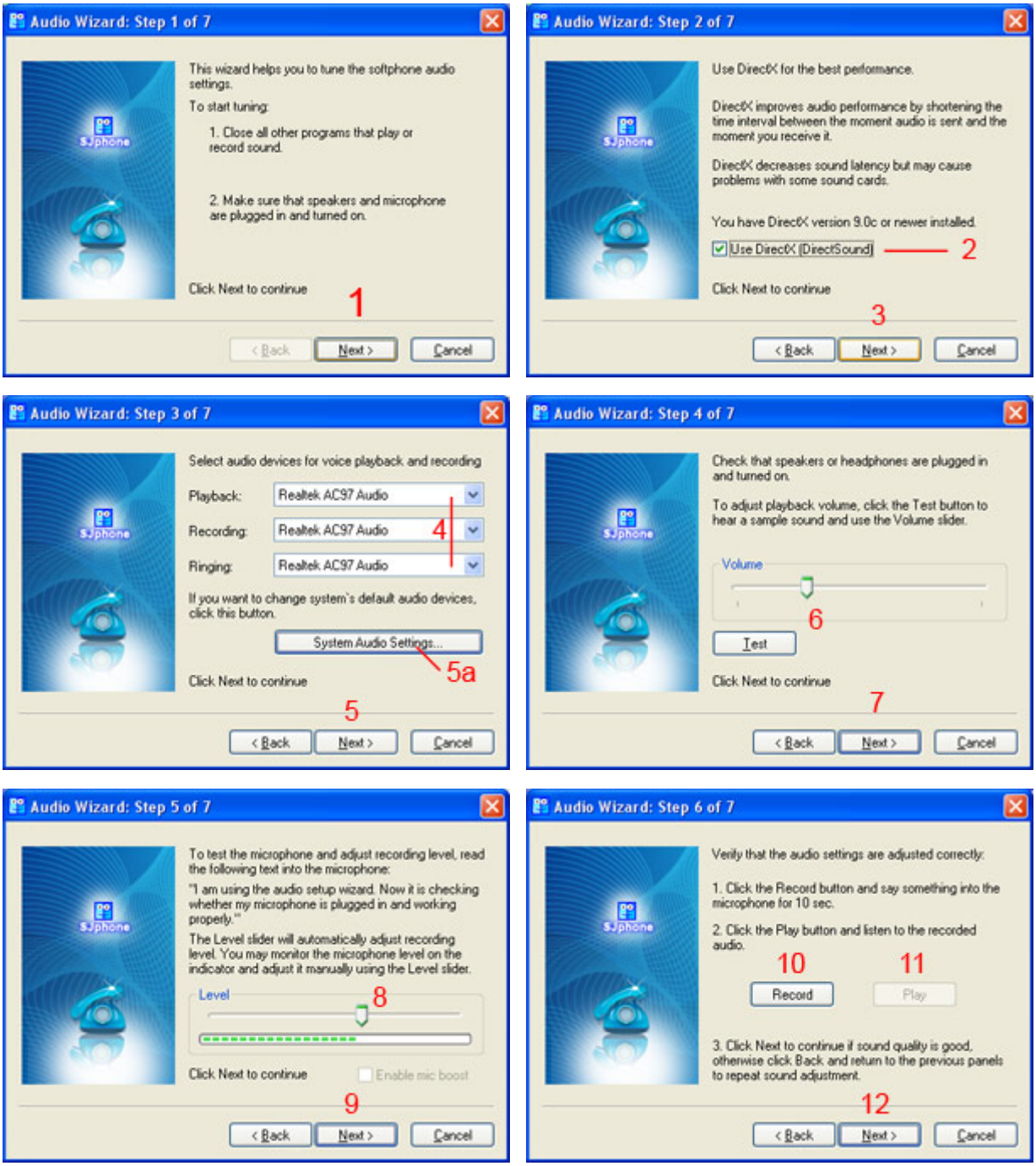


Fig. 4.36 SJphone Audio Wizard Screens 1 ~ 6

There is one more screen than shown above: the "Finish" screen. It's obvious so not shown.
The setup steps are detailed with notes over ……

SJphone Audio Wizard continued :

1. On screen 1 : Close any audio apps, and click [Next].
2. On screen 2 : Direct X is recommended - if you don't have it installed recommend you download and install to get a good result.
3. On screen 2 : Click [Next].
4. On screen 3 : Select your preferred system audio device for the softphone.
5. On screen 3 : Click [Next]
5a. On screen 3 : as mentioned previously you may need to come back here to set Windows audio device settings - see below.
6. On screen 4 : Using the slider and Test button adjust your playback volume to a comfortable level. Remember you can come back (or use system settings) to adjust it later if required.
7. On screen 4 : Click [Next]
8. On screen 5 : This is where it can get a bit tricky. The Mic volume is supposed to automatically adjust. I found it looked like it did but it really didn't. It was from this point and from the next screen that I found I had to go back to 5a and system settings to get adequate Mic volume - then upon return this auto feature didn't appear to work. Also the "enable Mic boost" tick box was not enabled for me - had to use systems settings to do that which is the process detailed below.
9. On screen 5 : Click [Next]
10. On screen 6 : Click the Record button and speak into your Mic. At normal conversational level for 10 seconds.
11. On screen 6 : Click the Play button and listen to the result. If not satisfied go back to step 8 and adjust the level and try again.
12. On screen 6 : Click [Next], you will be presented with the last screen of the wizard to which you can click [Finish].

Now the acid test will be when you have followed through the next section and set up the phone SIP settings and can make some calls. The most likely problem you may encounter is inadequate send (Mic.) volume. If you haven't got another extension close by where you can monitor your output you will have to ask other parties for feedback.
Say your Mic volume is too low : open the Audio Wizard and use the following as a guide.
From Step 5a above : Click the [System Audio Settings] button :

5b. System Audio Devices Properties : For Mic. volume click on the Sound Recording [Volume] button - you will get the Recording Control dialog.
5c. Click the [Advanced] button.
5d. On the Microphone Advanced Controls : tick the Mic Boost box to set boost ON.
5e. On the Recording Control : balance the Mic volume using both master and Mic control Go Back to the Audio Wizard (point 6) and work through again - you can leave the controls below open until you have the right levels.

If the Recording Controls doesn't show a Microphone tab then click on the Options menu and select Properties - then in the following dialog - tick the box to "show" the Microphone control.

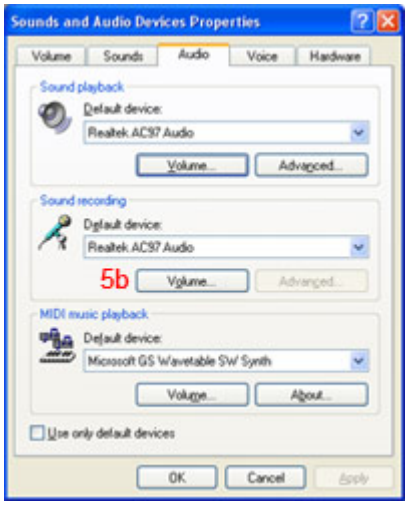


Fig. 4.37 Audio Devices Properties

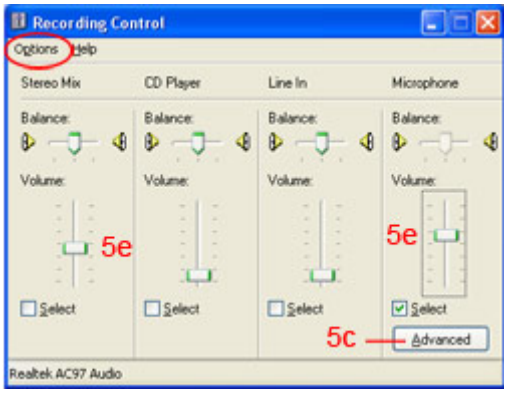


Fig. 4.38 Windows Recording Control

Fig. 4.39 Microphone Advanced

SJphone Audio Wizard continued :

In the case of some unbearable noise in the earpiece (speakers) other audio channels in Windows are prone to causing such and need to be controlled or muted:

5f. On the Audio Properties dialog you got from 5a : Click the Playback [Volume] button. This will bring up the Playback Volume controls.

5g. On the Playback Volume controls "Mute" or otherwise control all channels other than Wave and Mic. You will soon find the source of the interference. Adjusting these controls can be done at any time by accessing the Control Panel > "Sounds and Audio Devices applet.

Fig. 4.40 Audio Devices Properties

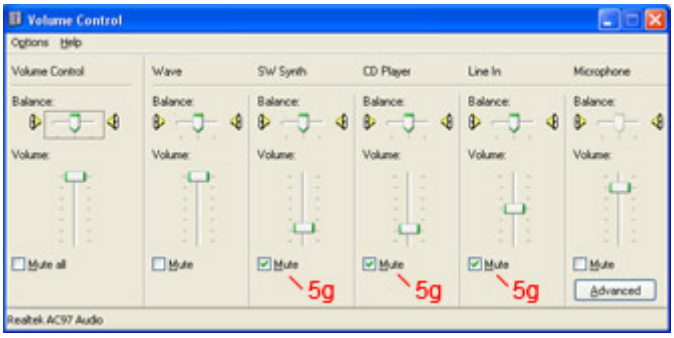


Fig. 4.41 Audio Playback Volume Controls

Just a note here : that the [Advanced] button on the Audio Devices dialog shown above will usually allow setting of specific speaker types for playback - you may wish to try adjusting these for effect.

SJphone SIP and Other Parameters Configuration :

After you have finished the Audio Wizard SJphone will usually launch - showing the first screen at right.
If it doesn't it can be started from the Systray or Start > Programs.

So - ready to make calls ? Not yet : the SIP parameter setups have to be done before you can do that.

Click on the "Menu" link at the base of the phone window. A Menu overlay will pop up. Slide your cursor up this overlay until the "Options" link is selected and left click this link.
See Fig 4.43.

The Options configuration interface will be shown set at the User Information tab

Fig. 4.42 SJphone Interface

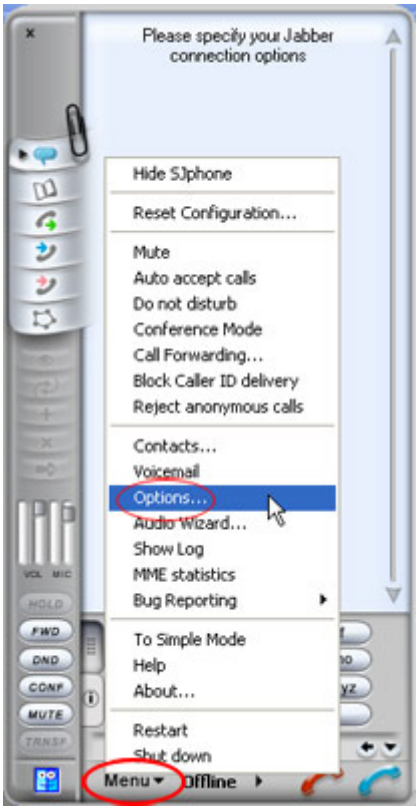


Fig. 4.43 SJphone Menu > Options

SJphone SIP and Other Parameters Configuration - cont'd :

From the Menu > Options link the Options dialog will be shown with the focus on the User Information tab. In the following figures all filled fields are shown as example only.

1. User Information : Fill in your details if you want.
2. Then click on the Profiles tab.
3. On the Profiles tab - Click [New] for a new profile.

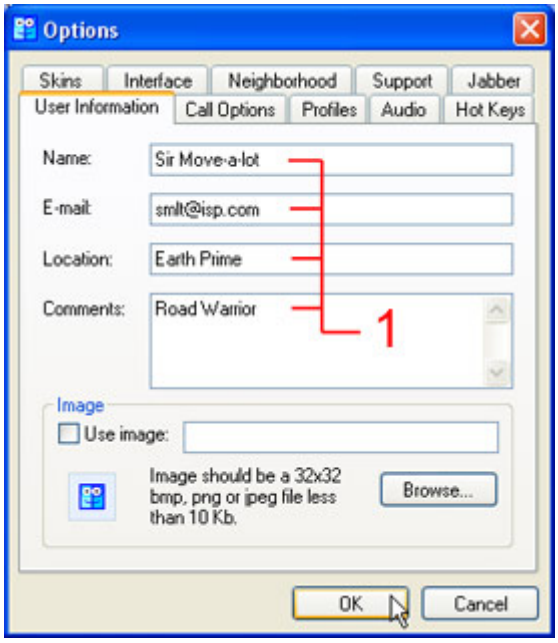


Fig. 4.44 SJphone Options - User

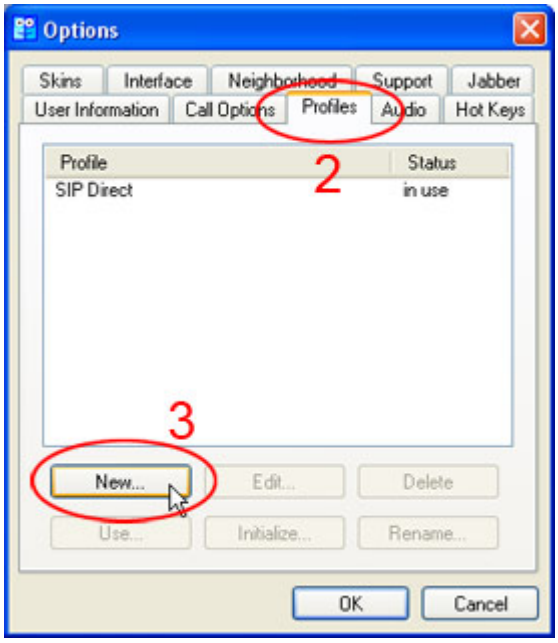


Fig. 4.45 SJphone Options - New Profile -

4. Give your new profile a name (example only shown)
5. Click the ellipsis and from the drop menu select "Calls through SIP Proxy"
6. Click [OK] - Profile Options now shown.
7. Click on the Initialisation tab and check the boxes as shown.
8. Click the Advanced tab - tick the boxes as shown.

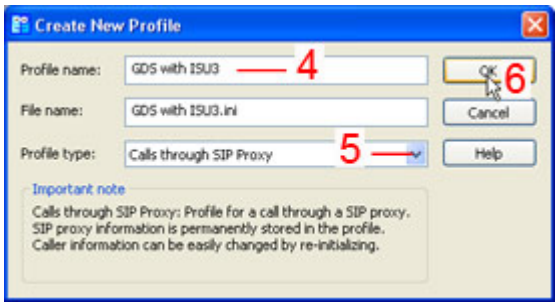


Fig. 4.46 SJphone Options - New Profile -

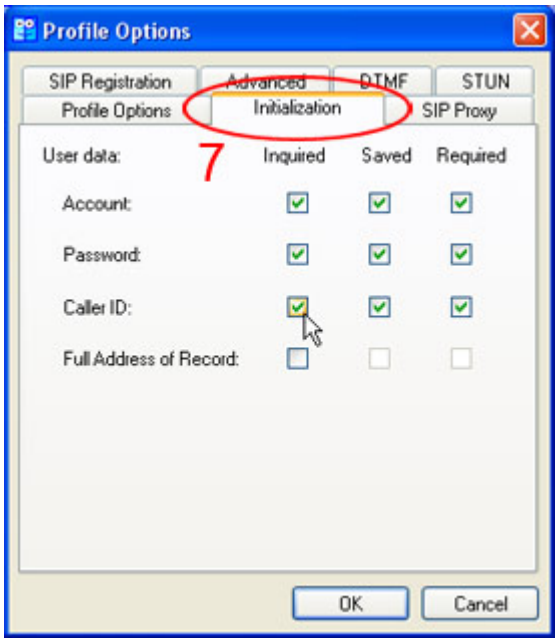


Fig. 4.47 SJphone Profile Options - Initialise Tab

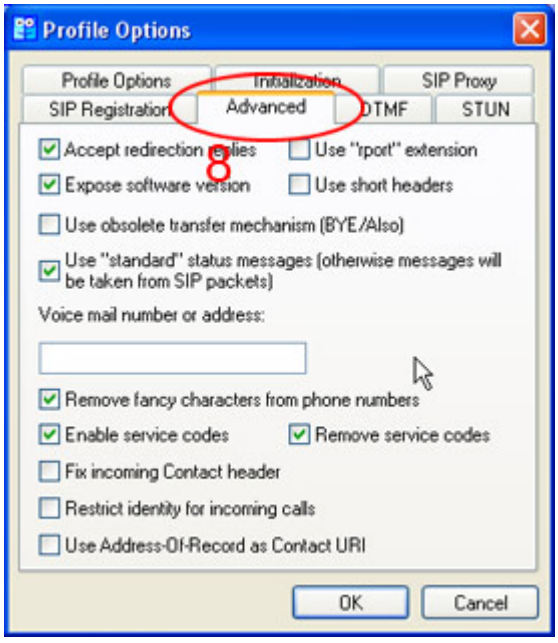


Fig. 4.48 SJphone Profile Options - Advanced Tab

SJphone SIP and Other Parameters Configuration - cont'd :

9. Click the SIP Registration tab - tick the boxes as shown.
10. The Suggested Expiration is the re-register time - like Options Interval Timer on ISU3 set this to under the router port bind time - suggest something less than 180 seconds.
11. Registrar URI is the URI of the account that you are setting up :
set this to the [GDS extension number you are going to use]@[ISU3 WAN IP]
(for ISU3 WAN IP see pt1, Fig 11, ISU SIP Proxy).
12. Click the SIP Proxy tab.
13. Domain/Realm : is the Registrar Server URI or IP - set to the ISU3 WAN IP
14. Proxy URI : is the Proxy Server to send call traffic to, prefixed with the protocol in use. Tick the "Use Outbound Proxy" box to enable the field then set to SIP:[ISU3 WAN IP] .

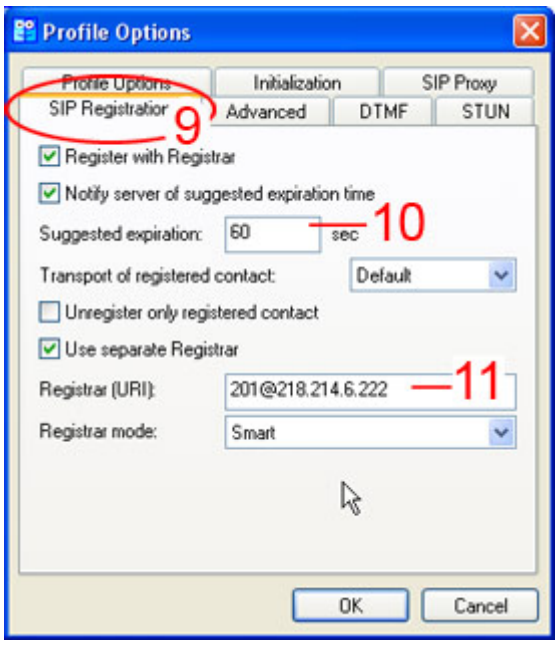


Fig. 4.49 SJphone Profile Options - SIP Registrar

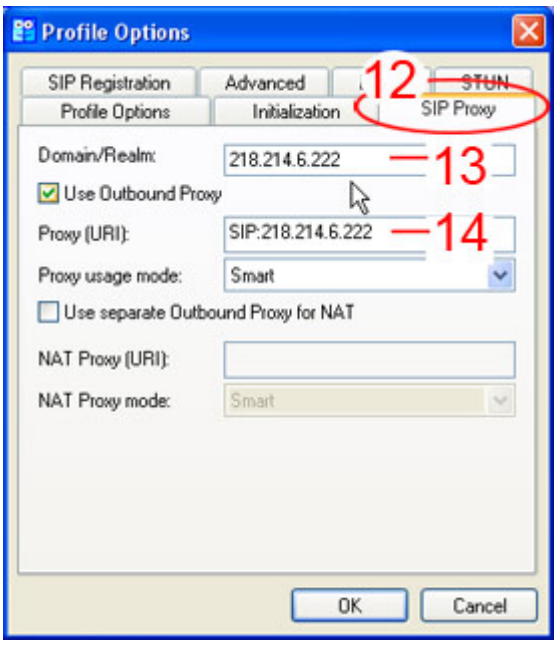
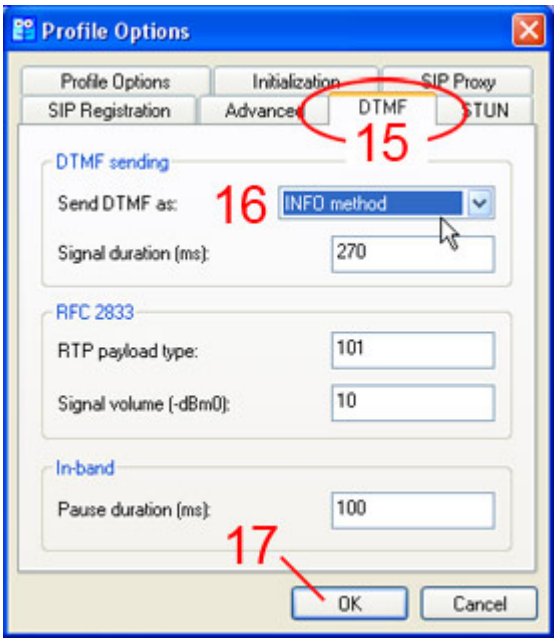


Fig. 4.50 SJphone Profile Options - SIP Proxy

15. Click the DTMF tab.
16. Using the ellipsis select "INFO Method" - this is the SIP Info style of DTMF we require.
17. There are no other tabs we need to fill now so Click [OK], and the service will initialise.
18. After 17 the Initialise dialog is shown - fill in your account details :
Account=[GDS Ext No], Password [doesn't matter], Caller ID=[GDS Ext No.]
19. Click [OK] to finish the initialization.



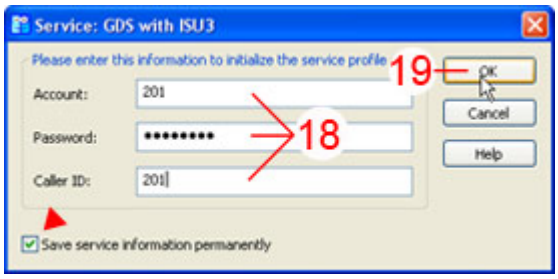


Fig. 4.52 SJphone Profile Initialisation dialog.

Fig. 4.51 SJphone Profile Options DTMF settings

SJphone SIP and Other Parameters Configuration - cont'd :

After you have initialised your profile you should be back at the Options dialog.
If not then select Menu > Options on the SJphone interface like before.

20. Click on the Audio tab
21. On the Audio tab : click on the Compression Settings button.
22. On the Compression Settings dialog select the G711a codec by clicking in the listing.
23. Use the [Up] or [Down] button to push 711a to the top of the list.
Then go back and select the G711u codec listing and push it to second in the list in the same way. These are the only two codecs compatible with our ISU3.
24. Click [OK] to exit the Compression Settings dialog.

Fig. 4.53 SJphone Options dialog - Audio tab

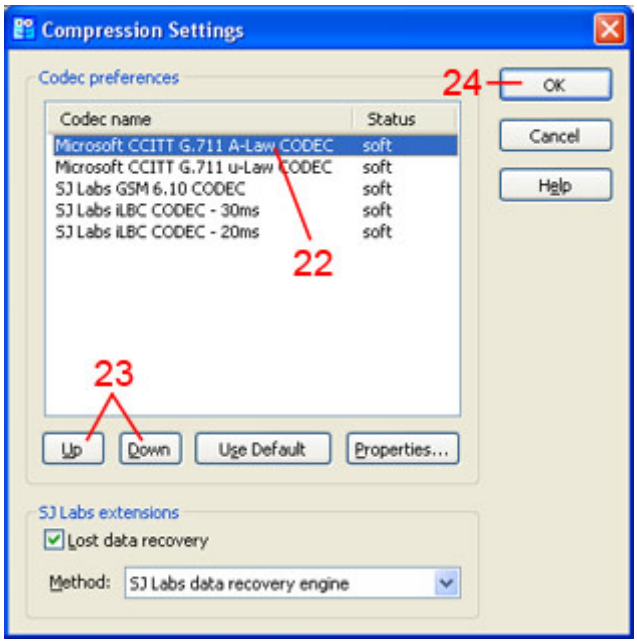


Fig. 4.54 SJphone Options - Audio - Compression Settings

25. Back on the Audio Options tab - click on the [Advanced Settings] button
26. On the Advanced Audio tab set the sample rate to 8000 (Hz).
27. Tick the boxes for Do Not Send Silence, and Echo Cancel Enable, **28**: [OK] to exit.
29. Back on the Options dialog you can select the Profiles tab.
30. The profile you have just made will be "in use" - meaning any calls you make will go out via this profile's Proxy etc. This is also where you set the "active" profile.
31.Click [OK] to exit and you should be ready to try making a few calls.

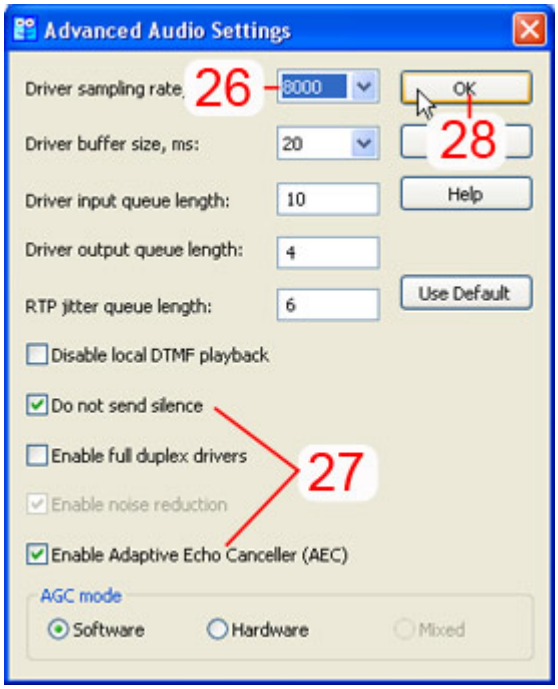


Fig. 4.55 SJphone Options - Audio - Advanced Settings .

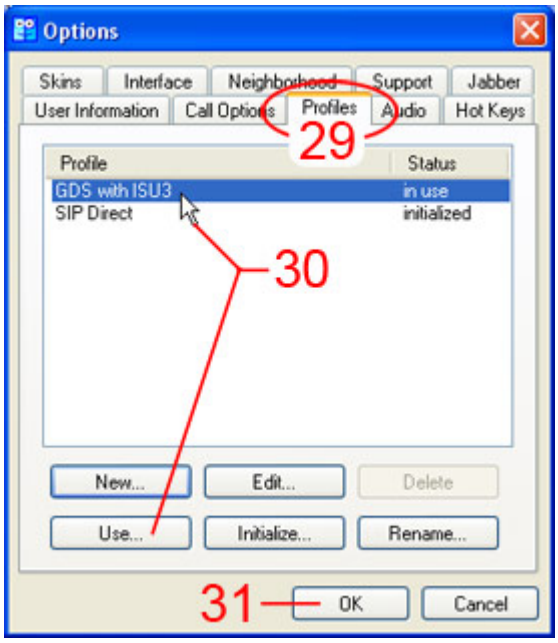


Fig. 4.56 SJphone Options dialog - Profiles tab.

SJphone Notes :

Once you have installed SJphone you will find it hiding in your Systray (lower right end of the taskbar) waiting to accept incoming calls. Double clicking on the Systray icon will pop the SJphone window onscreen ready to dial. Dialing, as noted previously, will send calls out via the settings of the “active” identity. If you have more than one identity installed you will know what you are doing and probably not be reading this.

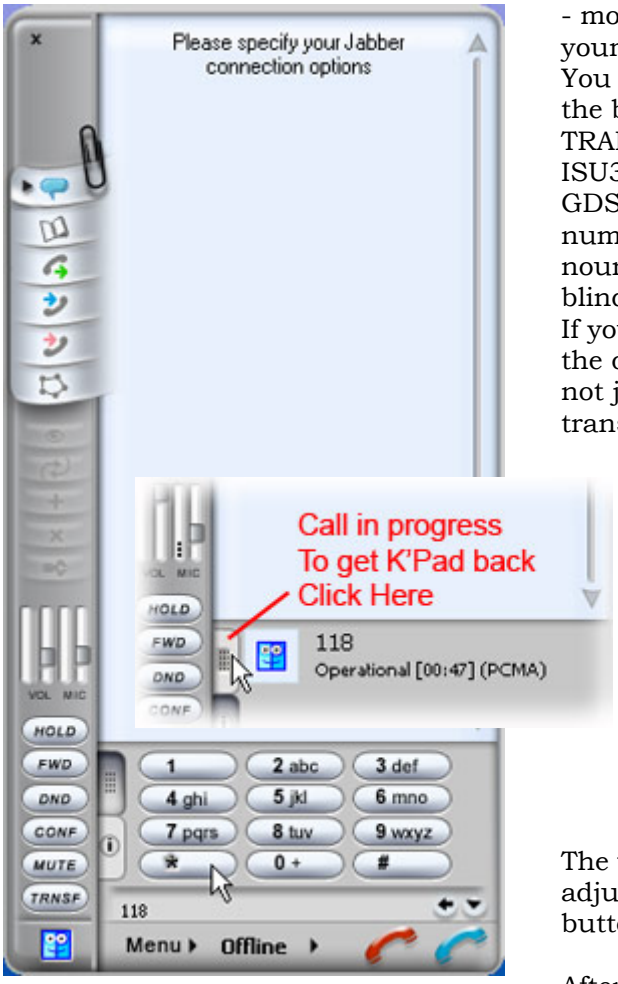


Fig. 4.57 SJphone Main Window

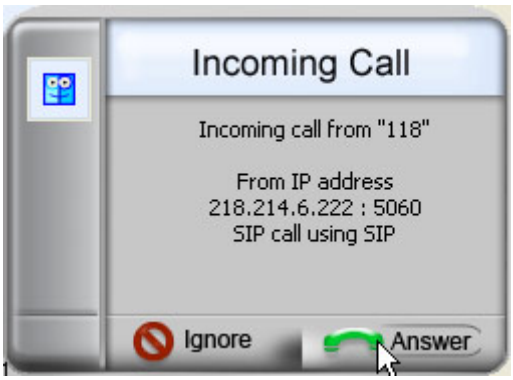


Fig. 4.58 SJphone Incoming Call pop-up

If you are new to SJphone you will need to explore the interface to find out where everything is - most areas will provide a tooltip if you hover your cursor over them.
You will find with the current programming that the buttons labeled CONF, DND, and particularly TRANSF are not functional. Call Transfer on the ISU3 is done by putting the call on hold at the GDS by dialing *# then dialing the extension number you wish to transfer to; then either announcing the transfer then hanging up; or doing a blind transfer by just hanging up.
If you do not wish to transfer the call then pick the original caller back up by dialing *# again. Do not just hang up as the original caller will be transferred if you do.

Dialing can be done from your PC keyboard, with the Enter key to send, or by point and click on the onscreen keypad. Once a call connects the keypad will often disappear so should you need to transfer a call from the GDS you can again use your keyboard to dial *#, or alternately there is a keypad button just to the left of where the onscreen keypad was - click this and the keypad will return for you to use.

The various SJphone parameters are available for adjustment or setting at any time from the Menu button.

After installing SJphone you will notice it load when your PC boots, then reduce to the Systray. As SJphone is resting in your Systray any incoming VoIP call will trigger an “Incoming Call” popup in the centre of your screen, along with the SJphone window which will be positioned where you last moved it to. After a call is completed the SJphone window will remain onscreen showing recent call records - including missed calls - until you hide it.

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Section 5

Routers

Contents

Modem Routers : Introduction

To connect any type of broadband network together, beyond a few local devices, a router is required. Where the network path involves transitions between the different transport mediums - Ethernet to/from ADSL for example - a modem is also required.
These two functions are often combined into the one device for obvious reasons.
As with other third party equipment, like the SIP endpoint devices, every manufacturer approaches methods and solutions for achieving device functionality in different ways and with differing interfaces. The functionality provided extends from basic to extensive, and is generally reflected in cost, with many different brands and devices available

For our ISU3 and IP extension implementation, particularly for the most suitable NAT'd ISU configuration, certain core functions are necessary. Such as port forwarding and redirect, which can be provided by most commonly available mid class modem routers. There is one function however that is provided by very few devices and that is the ability to carry and service IP extension devices on the same LAN that the NAT'd ISU3 is situated on. For reasons such as this requirement, and the ability to know and provide support to you the integrator for this most important piece of equipment, a number of devices were tested for function and a common choice of device and family settled upon which is detailed following.
This is of course for the installation at the "headquarters" GDS site carrying the ISU3/s. Whilst almost any router will suffice for the remote "extension" sites, and this device can provide port redirect for setting up desirable remote access/support to SIP extension devices such as Snom extension phones if that is mandated, it is felt that familiarity with and economic supply of the designated router/s below will see this device as the preferred alternative for this location also.

Before proceeding with setup details - some background information:

With the rising popularity of VoIP the trend for most low to mid class modem/routers is the inclusion of VoIP FXS ports along with many other feature integrations in an effort to draw market share with devices that do "everything". For those people just wanting to get cheaper phone calls from home by linking to a VSP like Engin or MyNetFone this is fine. For integrations such as our ISU remote IP extensions, and many other VoIP and/or IP-PBX style integrations, the inclusion of these features causes no end of trouble. Not because they are there but because of the way the various manufacturers have implemented the functionality. In most cases where a router has inbuilt VoIP ports the way that router handles SIP traffic means general SIP VoIP functionality for other devices behind that router is broken. This can sometimes be rectified with bothersome workarounds; sometimes the inbuilt VoIP ports have to be turned off altogether; sometimes there is no remedy. The point is, unless the special case exists where the noted inbuilt VoIP ports are being specifically used as the only IP extension endpoint for the network that router services, the use of routers with inbuilt VoIP capabilities is to be avoided.

If you are charged with organizing the broadband supply, and your choices are not limited by the particular location, a good place to start is the Broadband Choice website : http://bc.whirlpool.net.au/ . This site is the source of a great deal of information on this topic. Regarding the type of DSL supply the trend today is for ADSL2/+ due to the perception of massive speed advantages. Unfortunately for VoIP planning ADSL2 has gross variations in available speed depending on location and time of day. Straight ADSL is more reliable for the purpose of VoIP. Also "business" plans are favoured as more reliable and in some cases mandated as a "Static" IP is necessary for the ISU site. Data allowances are another consideration to be aware of : your data needs can be initially estimated using a formula based on anticipated talk time and codec choice. Say you use the G729 codec (30kbps/sec/channel) and anticipated voice use is 2 Hrs/day on each of the three ISU3 channels, data requirements would look something like : ((((30k/8) x 60 x 60) x 2)x 3) = 81 MBytes per day voice plus a notional 20% network overhead = 97.2MB/day or ~~2.2 GB per month (22 wkg days). For extensive deploys an ISP that provides end to end QoS via Diffserv is a distinct advantage. Beware of plans where uploads are counted - you'll pay double (e.g. some Bigpond plans). Finally plans such as "Naked DSL" are not recommended - a landline is advantageous especially in remote locations in case of 'Net outage and/or Emergency Services mandate.

Supported Modem Routers

The supported Modem Router for ISU3 installs is one of the Draytek Vigor devices.
For simple setups where the ADSL account is dedicated to the voice link and any of the network config types 1 thru 4, but not including VPN type, is undertaken the recommended device is the Draytek Vigor 2700e. There is also a WiFi enabled variant available - the 2700Ge - should WiFi AP be required. This device is ADSL/2/2+ enabled so can be used on the common DSL feeds available.
For more complex setups, where VPN's are required or multiple subnets are to be routed for example, the recommended modem router for use is the Draytek Vigor 2800, or 2800G if WiFi is required. This device has all the same base properties as the 2700e, and the same, but functionally extended, interface so the following setup details also apply.
Note - how to set up VPN's is beyond the current scope of this document.
The Draytek Vigor 2700 could also be used but for our uses the 2700e will suffice.
There are VoIP models (see previous notes) available in the Vigor range - denoted by the suffix "V" to the model No.- these should not be considered unless there are specific reasons for doing so as mentioned.

Fig. 5.1 Draytek Vigor 2700Ge shown 2700e is same without aerial

Fig. 5.2 Draytek Vigor 2800G shown 2800 is same without aerials

Draytek Router Setup

At this point the presumption is : that you have an ADSL account supplied; that you have the necessary details of the account such as login username and password, and the ADSL connection details if they differ from the norm; that your wiring is in place and you have a central ADSL filter fitted at the feed point (a central filter is mandatory on any ADSL2 feed, and makes life easier for std. ADSL feeds) since the POTS line carrying the ADSL will normally be connected to at least an SLT handset if not other equipment; and that you have a notebook or other PC and some Ethernet patch cables for the job.

For setup the usual method is to set the ISP login and LAN parameters offline, then put the router online and connect to the ISP, then continue with the rest of the setups.
The default LAN IP address of Draytek routers is 192.168.1.1. You will need to connect to the routers Web interface via this address so if your PC IP address is 192.168.1.xxx proceed, otherwise set your PC to "Obtain an IP Address Automatically" (instructions on IP address setting are in the Appendices if you need them).
Connect the router to its power supply and plug that into the mains, then wait a minute or so for it to boot. Then connect a standard Ethernet patch cable between the LAN port on your PC and one of the RJ45 switch ports (P1 to P4) on the router.
Open a browser on your PC and put the following into the address bar http://192.168.1.1
Click Go - or tap the Enter key - and you should see a login window appear on your display.
The Draytek routers **default login username is admin**, there is **no password**, enter these and click [OK]. The routers Web interface will then be shown (see over) If you do not obtain the login window check your connections and PC's address (see Appendices). Once logged on you will also need Java installed and enabled for the router interface to work properly.

Draytek System Status Page:

Continuing from the previous page the first Web console page shown by the Draytek will be the System Status page :

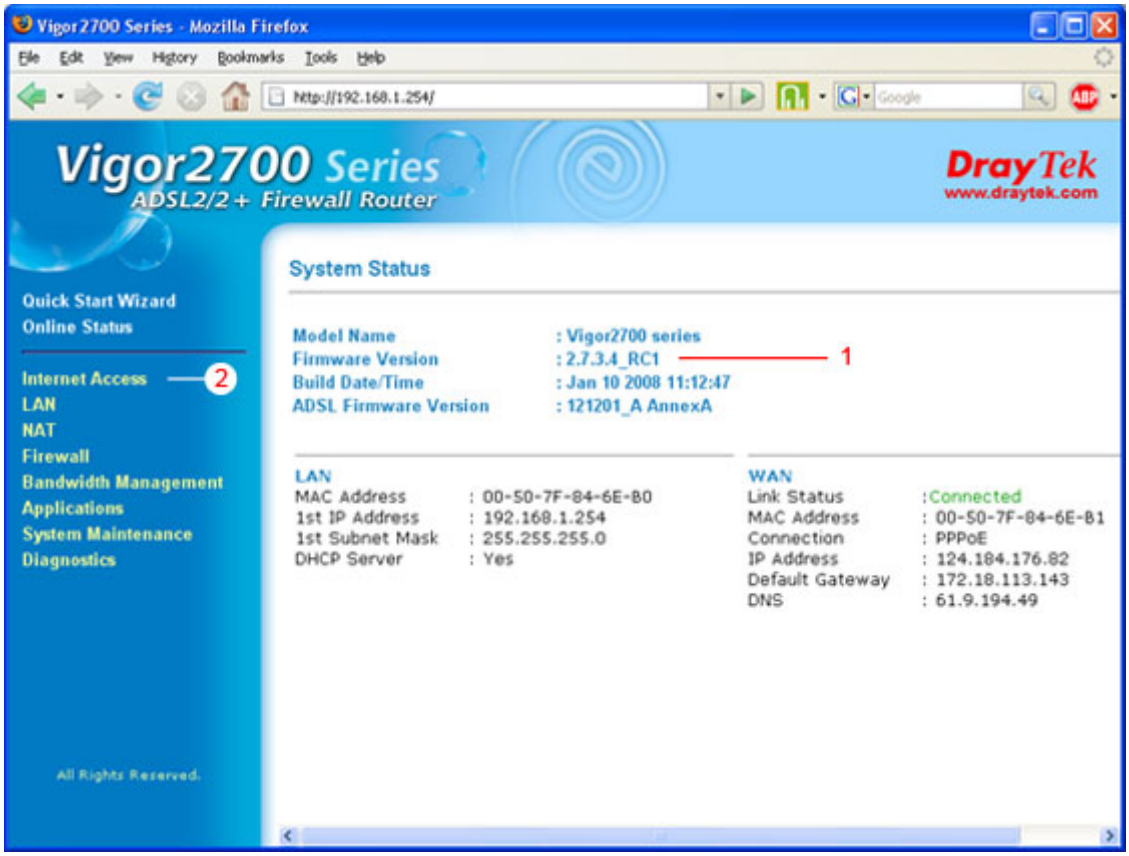

Fig. 5.3 Draytek Vigor 2700e System Status Page (example only)

The System Status page shows the relevant details of the modems status:

1. The modem firmware version.
   For a 2700Ge/e modem this version No. must be at or later than that shown.
   The latest versions of firmware can be found here: http://www.draytek.com.au/downloads.php. If you need to upgrade there are instructions on the site. You need to download and install the "Router Tools", as well as download the firmware version from the correct link. If the version number your device shows is close to that required then you can proceed with the setup and download what you need after you get online. Upgrade can then generally be achieved without losing any settings provided you use the ".all" version of upgrade firmware.

The interface menu is shown on the left of the page - if you can't see this then you either don't have Java installed or it is turned off - you will need to remedy this before proceeding.

The figures below the horizontal divider show the interface details.

Please Note : The page example shown above, and those following, are for a router that has been set up and is online. The details such as addresses etc are only an example and will be different to the device you will be working with. So the only settings to be observed closely are those that are indicated or by instruction.

2. Click on the Internet Access menu item - then click on the PPPoE/PPPoA menu item shown in the expanded menu drop. The PPPoE/PPPoA page will be shown - over : ..

Draytek Internet Access Parameters Page :

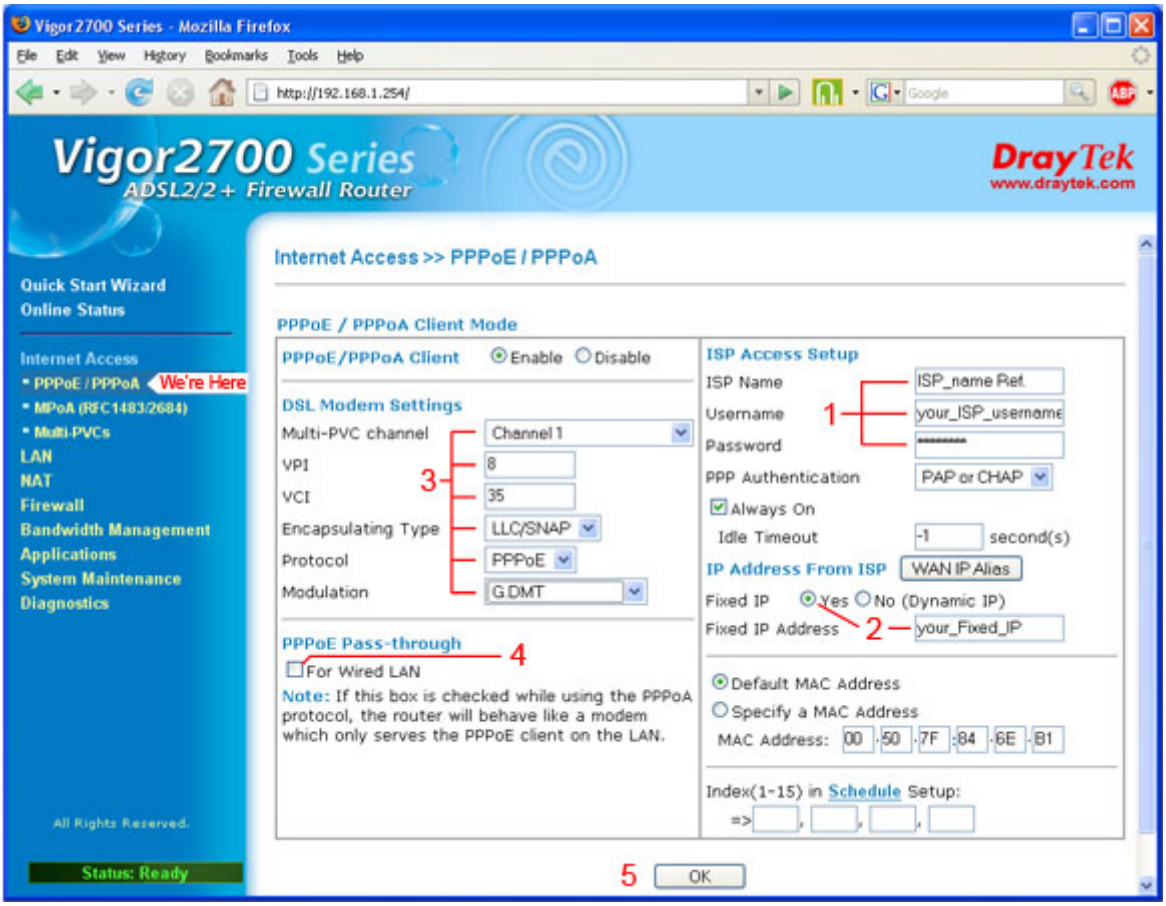

Fig. 5.4 Draytek Vigor 2700e Internet Access PPPox Page (example)

The Internet Access >> PPPoe/PPPoA page allows setting of ADSL and ISP login parameters :

1. Enter the relevant details here : the ISP Name is for your reference only; the Login Username and Password will have been provided by your ISP. Note the login username and particularly the password are case sensitive.

2. Since a prime requirement is an account with a Fixed (Static) IP this will also have been noted by your ISP. Enter it here and tick the Yes radio button
This step is not mandatory as the IP can be observed after connection is obtained but experience is that successive logins are easier with the entry made.
If a static IP is impossible to obtain there is another (less desirable) option - see the Appendices note on DynDNS.

3. The parameters of the ADSL connection need to be set here. A common ADSL form is shown. If required : the correct settings for these parameters will either be notified by your ISP in the account commission completion documents, or be available on the ISP's Website, or by telephoning their support lines.

4. If you are setting up a single ISU3 in PPPoE mode (Network Type 3 see ISU3 sect pp7) you need the Draytek to act as a modem only - tick this box. The DSL modem settings will still matter, the ISP Access Setup section (1) will not, nor will most, if not all, of the pages following in this section. See also ISU3 PPPoE mode detail ~pp12

5. Don't forget to click OK to save your entries before leaving the page.

The next step will be to set the routers LAN address, which determines the address range of the LAN subnet, and other factors such as DHCP if required. See over ..

Draytek LAN Parameters Page :

On the router menu : click the LAN item. Then click the General Setup item in the expanded menu drop. The LAN settings page will then be shown :

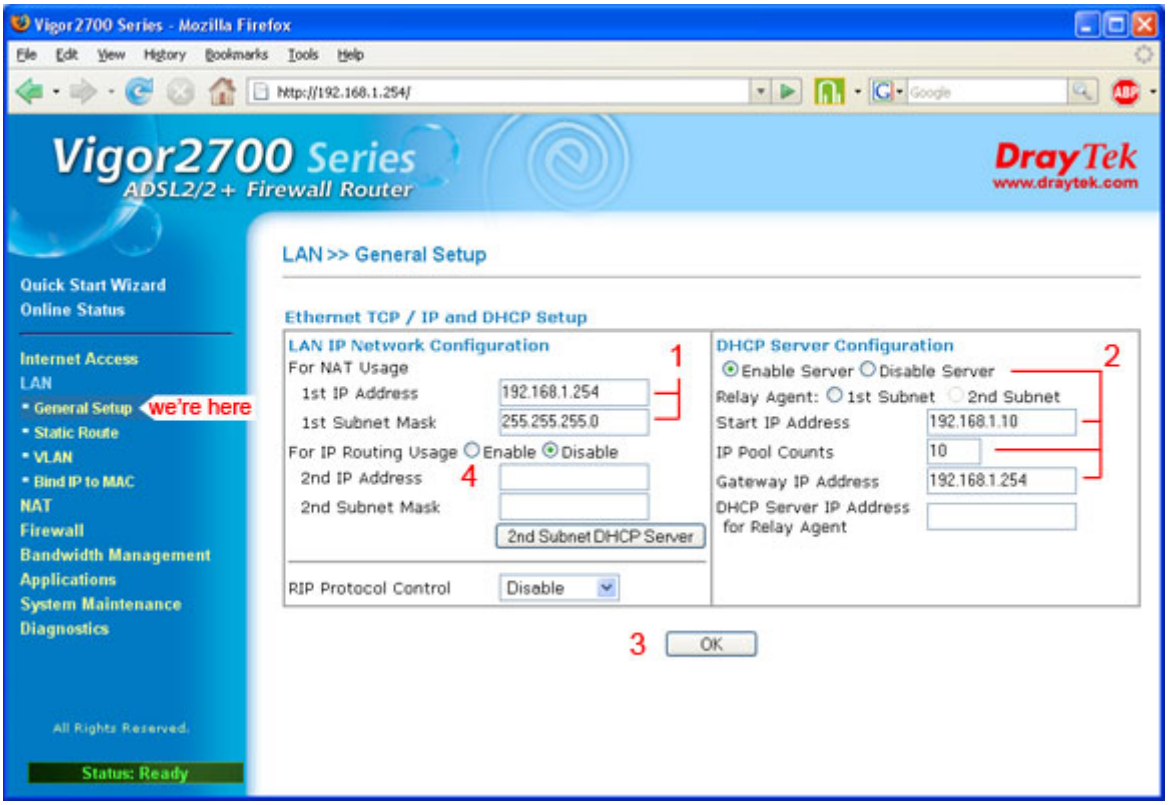

Fig. 5.5 Draytek Vigor 2700e LAN >> General Setup Page (example)

The LAN General Settings page allows you to configure the router's LAN parameters if you wish or are required to conform to addresses other than the default. (the default LAN IP of Drayteks is 192.168.1.1, subnet is class c : 255.255.255.0)

1. Enter the desired LAN IP of the Router and relevant subnet mask here.
   This needs to be one of the private address ranges defined by the Internet Protocol standards which can be obtained by Internet search if required.
   The IP address entered here will be the "gateway" address for any devices situated on the routers LAN. The accepted norm is the gateway address is usually at the bottom (.1) or the top (.254) of the subnet range for a class C subnet like that shown - but this is not a mandatory rule. The subnet mask will also apply to LAN devices and will in fact determine the available addresses such devices can use.
2. If a DHCP server is required this function can be enabled here. As you might have noticed it is enabled by default - that is what allowed your PC to get an address if you had set it to "automatic" at the start. The "Start" IP address is the numeric point from which DHCP addresses issued start. The Pool Count determines how many address can be issued. The Gateway IP address will usually be set to the same as the routers 1st LAN address (1). There can only be one DHCP server on a network - if there is another on your network one of them must be turned off. Do not set any device to a fixed IP that is within the DHCP range (Start + Pool count)
3. Don't forget to click OK to save your settings before you leave. If you have made any changes - when you click OK the router will require a Reboot to apply the settings and you will then need to put the new router IP address into your browser address bar to log back onto it afterward.
4. Not immediately relevant here but provided as a matter of interest : the second LAN address setting is how you swing a routed public subnet onto the routers LAN unimpeded by the routers NAT functions, and thereby enable the assignment of public IP's to devices on the LAN. The public address assigned here becomes the gateway for devices that are assigned any of the other available public IP's in the routed subnet.

Draytek Online Status Page :

After following the setups on the previous pages the next step will be to put the router online. Presuming that the router rebooted after the last step, and that you have browsed again to the router Web interface and have it displayed on your PC :
Now connect the ADSL feed to the DSL port on the router.
On the router Web interface click the Online Status menu link. The Online Status page will be shown :

Fig. 5.6 Draytek Vigor 2700e Online Status Page (example)

Note: as before, and after, the page shown above is of a router that is configured andonline and is provided as example only, so the numbers you see on your unit will be different.

The purpose of using the Online Status page at this time is to observe the progress of the login. Your unit may login immediately so you will see something like the above - but usually the first login takes a while so you can observe the process as a guide to where there is trouble if the login is unsuccessful.

1. The first thing to happen is the sync of the DSL connection. The mode will show various words such as “Ready” …”Training” .. “T413” etc until the DSL sync is obtained. DSL sync is announced by the term “Showtime” as shown. The link speed is also then shown - although experience is that this is not to be taken as gospel.
   If sync is not obtained after say 3 minutes it is a safe bet that you either have your settings wrong : go back and check the PPP page; check with ISP for correct settings etc; try other settings; - or there is something wrong with the service (unlikely).

2. After sync (1) the PPP login process will take place. This is announced by terms such as “Starting PPP..” etc. When login is successful the mode will be displayed and the public IP assigned to your connection will also be shown. If the modem syncs but PPP login does not succeed within a couple of minutes - go back and check your login details - check and re-enter the password - be aware of case sensitivity - be also aware that if you read the password off the ISP website for example that some display fonts will show characters such as lower case “el” and numeral 1 as the same thing.

3. After login the ISP preferred Primary and Secondary DNS server IP’s will be shown for your reference.

Draytek Port Forward Settings for Type 2 ISU3 Network Config.

Now we have a router that is online so the account details are verified. If your PC IP address gateway is set to the address of the routers 1st LAN IP you might try browsing the Internet to satisfy yourself that all is well. The following instructions presume you are proceeding with a Type 2 network setup (NAT'd) for the ISU3.

Before continuing with setup - a brief note on "ports" :
The term port, for traditional telephony uses, is generally applied to a connection point eg trunk port, extension port, Ethernet port, etc. all of which imply a physical connection point like an RJ connector socket. In Internet Protocol (IP) parlance, as part of one of the protocol layers that make the complex task of managing data communications possible, the term "port" refers to reference numbers between 1 and 64000 - one for the source and one for the destination - that are communicated in each IP packet along with source and destination addresses etc. These port numbers identify distinct data streams and eventually therefore the program or process that is associated with that data stream and is therefore its handler.

The next step after the router is online is to connect your ISU3/s and set the port forwards. Presuming that you have set the IP address of your ISU3 (Network Settings) to be in the routers LAN subnet, the gateway IP agrees with the routers 1st LAN IP address, and the subnet mask is set appropriately : connect the ISU3 LAN port to one of the Ethernet ports of the router with a std Ethernet patch cable.
On the router Web interface click the NAT menu item, then click the Open Ports menu item from the expanded drop menu. The NAT Port Forward index page will be shown :

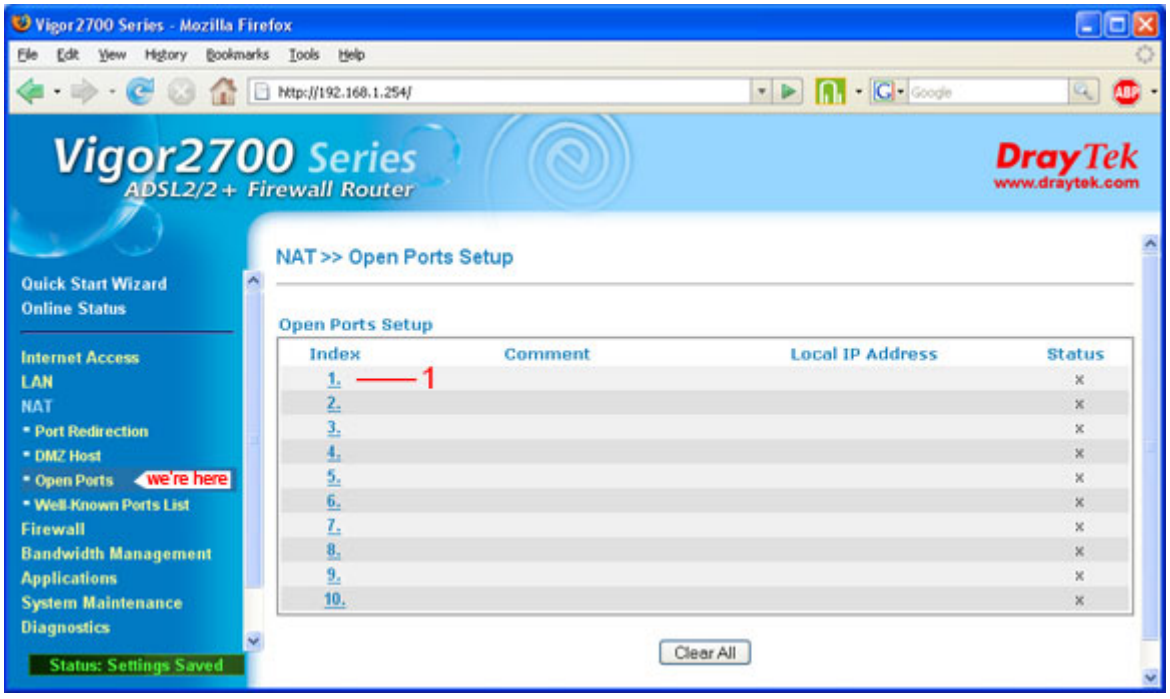

Fig. 5.7 Draytek Vigor 2700e NAT Port Forward Index Page (example)

The Open Ports Setup page shows an index of 10 port forward groups. These groups can each contain 10 port forward sets for each index entry - as will be seen on the following page. These groups are used for setting straight port forwards for single ports, or groups of ports, to a particular LAN device. A straight port forward is where the traffic on a particular port number on the WAN interface is forwarded to the same port number on the LAN device. Note that once specific port number/s are forwarded to a LAN device no other LAN device can expect to see traffic from that WAN port number - it is in effect consumed.

1. Click on the Index link to Edit the port parameters for that index group.
   The Edit page for that index group will then be displayed … over :

Port Forward Settings for Type 2 ISU3 Network Config cont'd :

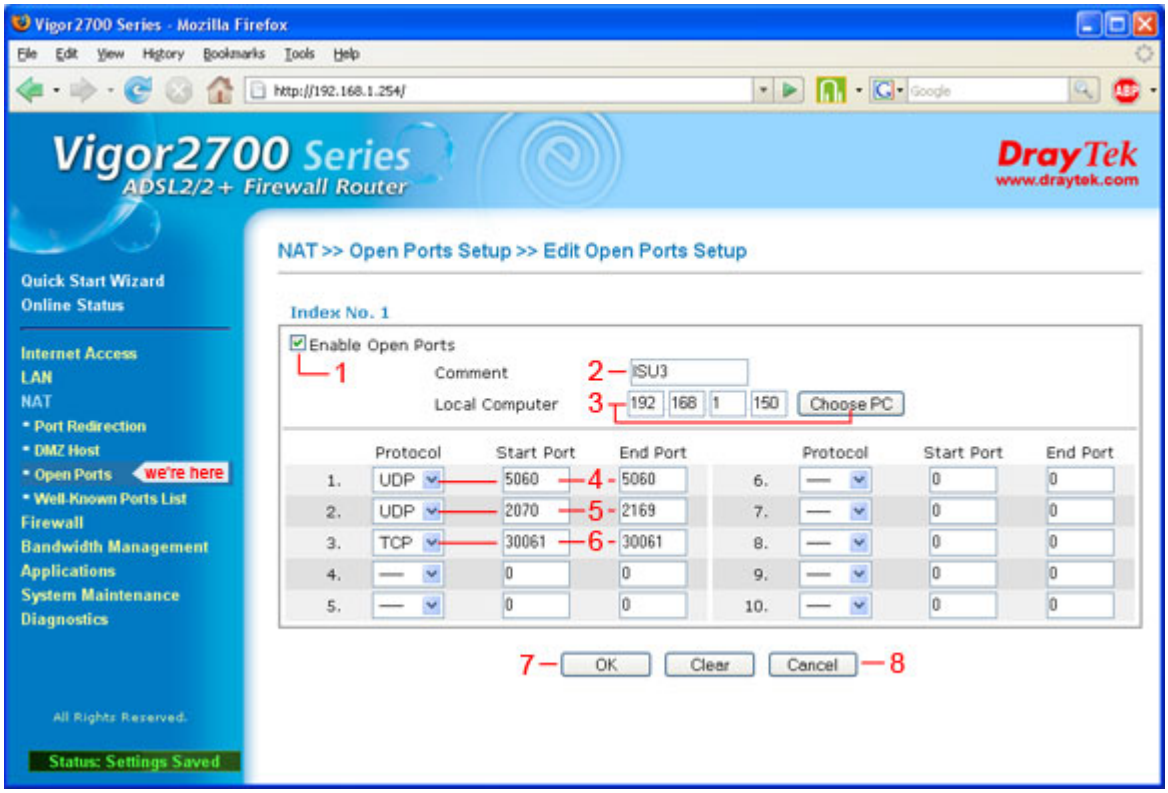

Fig. 5.8 Draytek Vigor 2700e NAT Port Forward Index 1 - Edit Page (example)

1. Click in the Enable Open Ports box to enable edit for this Index group.

2. Enter a name for the port forward group for your reference.

3. Either enter the LAN IP of the ISU3 (in this case) - or if the ISU3 is connected to the router and its LAN IP is properly configured you can click the [Choose PC] button : this will pop a small window onscreen showing the connected devices on the LAN from which you can select the correct device address by clicking on it.

4. 1st port forward : Set the protocol and the port numbers.
The same number in both Start and End ports signifies a single port forward.
This is for the SIP command porting so the protocol is UDP and the port number to start is 5060. Note as referred to in the ISU3 setup section if multiple ISU3's are installed then each has to be assigned a different port eg ISU3-1 =5060, ISU3-2 =5062

5. 2nd port forward : Set the protocol and port numbers.
In this case it is a port range, for the RTP voice channels, the protocol is again UDP.
Again Note - as referred to in ISU3 setup section if multiple ISU3's are installed then each successive ISU must be assigned a different port range for the RTP. For example ISU3-1 (RTP) = 2070 ~ 2169, ISU3-2 (RTP) = 2170 ~ 2269, etc.

6. 3rd port forward : Set the protocol and port numbers.
This port is TCP because it will be used to remotely access via a browser the ISU3 card this port set points at. The single port 30061 is set as that is the native Web I'face port of the ISU3. For any other ISU3's installed you will have to use the Port Redirect function (described next) to set WAN (Internet) access ports for, as the ISU3 Web I'face port No. is not configurable.

7. Click [OK] to save your settings - which will now be active by the way.

8. Since the page stays onscreen after the OK button - Click the [Cancel] button to return to the previous screen. Or you can use any Menu link to go elsewhere.

Draytek Port Redirection for Remote Access to Devices.

Port redirect is where incoming traffic on a particular port at the WAN interface is forwarded to a specific LAN device but in the process the traffic is redirected to a different port number on the LAN device. Limitation is : only single ports can be redirected, not groups.
For our purposes Redirect is useful for providing remote access to the Web browser interface of devices situated on the LAN of this router. Since the configuration of interest here is a type 2 (NAT'd) ISU3/s with a dedicated broadband connection it makes sense to utilise this connection to also provide remote programming and/or monitoring for the GDS carrying the ISU3/s by connecting the GDS to this LAN and opening (redirected) ports to it. Other instances of this redirect feature use is allowing remote access to any Snom phones on the LAN, or remote access to more than one ISU3 card in the GDS as noted previously.

An example : Click the NAT > Port Redirect menu link on the Draytek Web page :

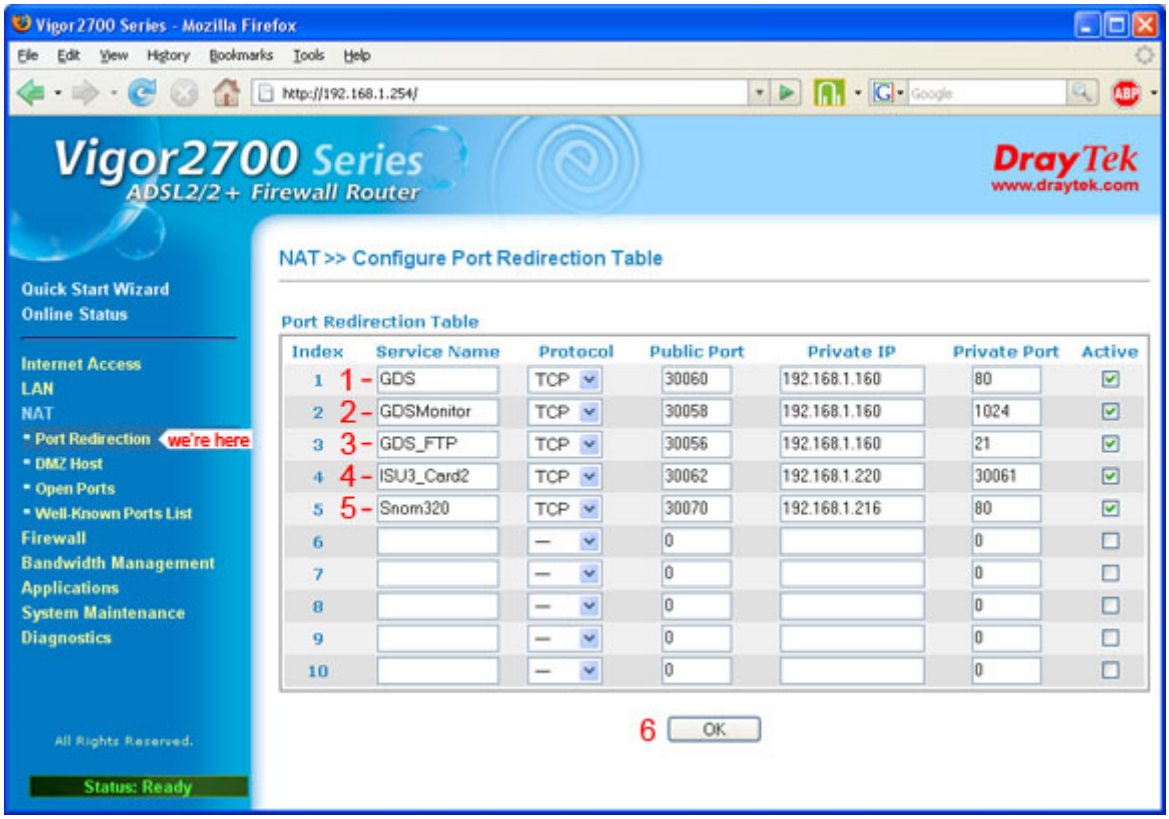

Fig. 5.9 Draytek Vigor 2700e NAT Port Redirect page (example settings shown)

Note before proceeding : any **Public Port** number set must not be used anywhere else on the WAN interface - in standard port forwards for example, or as a "common" port for any other program/process in the case of a mixed data network (which is not recommended btw).
The Public Port numbers shown above are arbitrary and shown as example only.
Service Name is for your reference only. All browser communications use **TCP** protocol.
Public Port is the WAN port, **Private Port** is the LAN device port
Private IP is the (fixed) IP address of the LAN device that is the target of the forward.

1. Port redirect for GDS remote programming - redirect is to GDS port 80 for Web interface.
2. Port redirect for GDS Monitor shown - redirect is to GDS port 1024 for GDSMon I'face.
3. Port redirect for GDS FTP interface shown - redirect is to GDS FTP port 21. Use for remote backup of system settings. Not recommended to do firmware upgrades over Web.
4. Example of port redirect for second ISU3 card remote programming shown - this redirect method is only necessary for the ISU3 Web console as the other differing port numbers for SIP and Voice traffic would be done via standard port forward settings.
5. Port Redirect for Snom320 IP phone remote access example shown.
6. Don't forget to tick the Active boxes on the right and click **OK** button before you leave the page so your settings are saved and activated.

Draytek Router - Remote Management

Whilst remote access to the ISU3 and other LAN devices is an advantage there are also advantages to allowing remote access to the Draytek router itself. Such as being able to observe the ARP cache (see notes) to check devices are connected, reform port forwards based on any IP changes etc.
There are caveats of course in that the changes you could make could also be made by a hacker if they are able to spoof your details and know your password. So along with the prelim defenses noted below it is *imperative* that you set a strong password on the router before enabling remote access. As noted before a strong password is generally regarded to be a non-dictionary term consisting of a range of alphanumeric characters that is at least 8 characters in length. (alphanumeric = numbers and symbols as well as letters).

To set up remote management : click on the Draytek's menu link :
System Maintenance > Management :

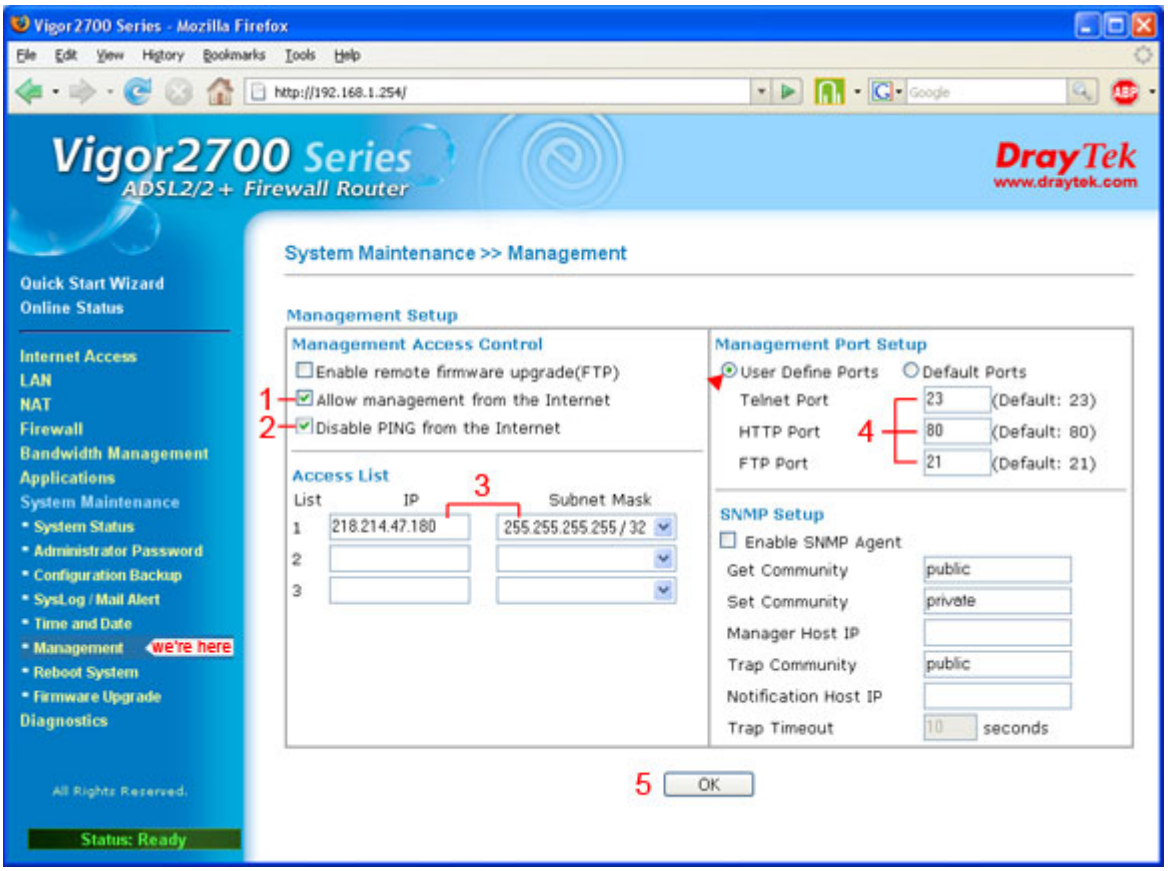

Fig. 5.10 Draytek Vigor 2700e Management page (example settings shown)

1. Tick to allow remote management via the WAN side (from the Internet).
2. Disable Ping means router will not respond to (Internet) ping requests - this is a first line of defense against hackers looking for available devices to attempt to exploit.
3. The router will accept remote management requests from traffic bearing the IP addresses contained in this Access List. Here you will enter the WAN IP of the broadband connections you will use to access the router. The subnet mask is set as shown for a single account - for a connection that has other WAN IP's in it's WAN pool - via a routed subnet for example - the subnet mask is set appropriately. (In fact the subnet mask can be set so wide that access could be obtained from anywhere (not a good idea).
4. Shown are standard port settings - these values can be manipulated as an additional line of defense or because the ports are forwarded for other purposes. Make a note in your install logs if you change any of these.
5. Click OK to save your settings - the router will then request reboot to apply new settings.
6. Go to System Maintenance > Administrator Password and set a strong password - and *please* note the new password in your install logs immediately. If you forget the password the penalty is a site visit incl. factory default the router and set up again.

Draytek Routers - Notes :

Firewall Features

You will notice the Vigor 2700e/2800 has a fairly extensive set of firewall features. In most cases for our use these settings can be left at default. The default call and data filters are basically only NetBios blockers so do not affect our traffic and for general security purposes will be left enabled. With the requisite knowledge you may wish to add other filters - if so it is recommended that you thoroughly test any changes. The DoS Defense page is another area that may interest you if you feel vulnerable to this kind of attack - but be aware some of the mechanisms available on this page definitely affect VoIP traffic so again test anything that you do change. This information is provided as background information only, not as supported setup detail.

Bandwidth Management

The Vigor 2700e has rudimentary bandwidth control, the 2800 has a more extensive set of features. In usual practice, since the connection is recommended as dedicated to voice, bandwidth control should not be necessary. The only observation is to check that the Bandwidth Limit is disabled if the ADSL connection speed is higher than the defaults shown. If however you have the situation where other non-voice devices are sharing the connection (hopefully this is only the GDS system) then one way to counter contention with voice data streams is to limit the non-voice devices to a bandwidth that ensures there will always be enough bandwidth available for the ISU/s. Be aware that this control will only be applied in the upload direction. Since it is anticipated that a GDS may be ported through this connection the highest bandwidth this would currently require (in the absence of CTI) would be if it were supporting a GDSMonitor stream for faultfinding: a casual observation of such a stream showed usage up to 10Kbps Tx (upload) with a fairly busy system.

System Maintenance

Apart from the pages described previously there are two other useful features in this menu. **Configuration Backup** allows you to save a backup of all the router settings to a file on your PC which is useful security against any setup alterations, or in the case of a router firmware upgrade using the '.rst' firmware versions.
The **Syslog** feature is also useful in some situations where closer detail of connections, traffic, user access, etc, is required. In this case the Draytek "Router Tools" (see www.draytek.com.au/downloads.php) are installed on a PC and the "Syslog" application in these tools is run; the router Syslog page parameters for the logging are entered with the resultant stream pointed at the PC doing the logging.

Diagnostics

There are a range of diagnostic pages available that are of cursory interest. A useful page from a support perspective is the **ARP Cache Table** which shows a list of active network device IP's as related to their MAC addresses. MAC addresses are the foundation of Ethernet communications as they uniquely identify the Ethernet port of a device. The first six digits of a MAC address identify the manufacturer - for example all current Hybrex SIP equipment shows 00-09-85-xx-xx-xx, Snom Phones show 00-04-13-xx-xx-x. So it can be seen when devices such as the ISU3 and Snom phones are online and the IP they are using. Be aware however that the ARP cache only shows devices that are actively using the WAN port, so a LAN connected GDS will not show unless it is being remotely accessed for example.

Applications

The only application that the Vigor provides that interests us is the **DynDNS** feature.
DDNS (or Dynamic Domain Name Server) is a facility that can be used to counter the lack of a fixed or static WAN IP. There are free services available on the Internet that will provide a DNS service to dynamic identities (IP's) provided any IP change is reported to them. The Vigor DynDNS application does just that : enter the details of the DDNS service provider and your account login username and password and at every reboot the Vigor will pass the (new) WAN IP to the DDNS service. A well know DDNS provider is www.dyndns.com.
The disadvantage of using DDNS for voice services, as opposed to a static WAN IP, is slower connection times and the increased chances of unreliability (one more link in the chain).

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Section 6

Appendices

Contents:

Appendix A :- How to change your PC's IP address :

To change your PC's IP address you need to access the Network Connections applet.
Depending on the operating system (we are talking about MS Windows here) there are a number of ways to do this. A common way is shown here for a Windows XP running in 2000 "Classic" mode. If this is not available to you will find other links for Network Connections in the Start menu. Failing all that : the Network Connections applet can always be found in the Control Panel.

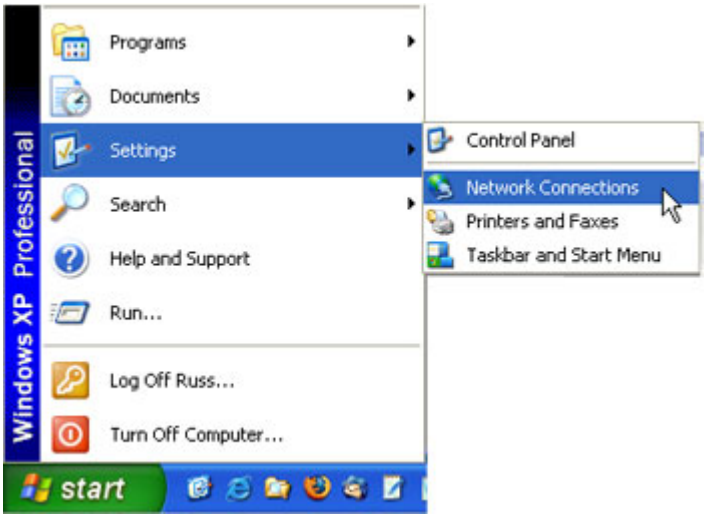

Fig. 6.1 START > Settings > Network Connections

Click START > Settings > Network Connections :

1. On the Network Connections Dialog:
Right Click on the active Local Area Connection icon and from the context menu drop select "Properties" :

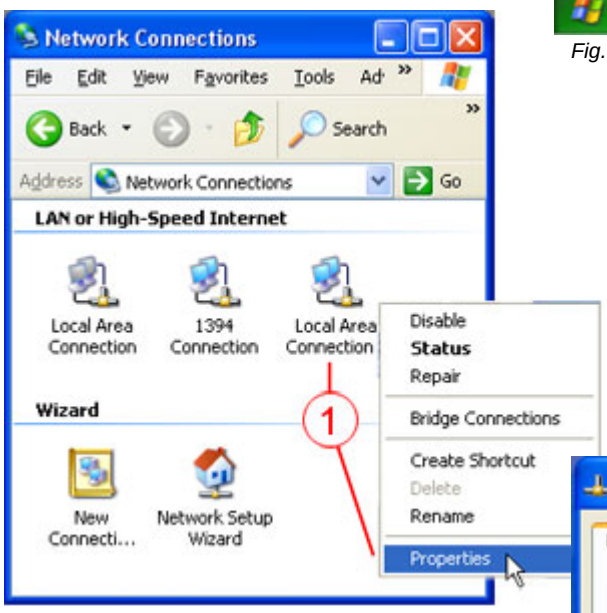

Fig. 6.2 Local Area Connection - Right Click > Properties

2. On the Local Area Connection Properties Dialog :
Scroll down the items
Select TCP/IP item by clicking on it

3. Then Click the Properties button

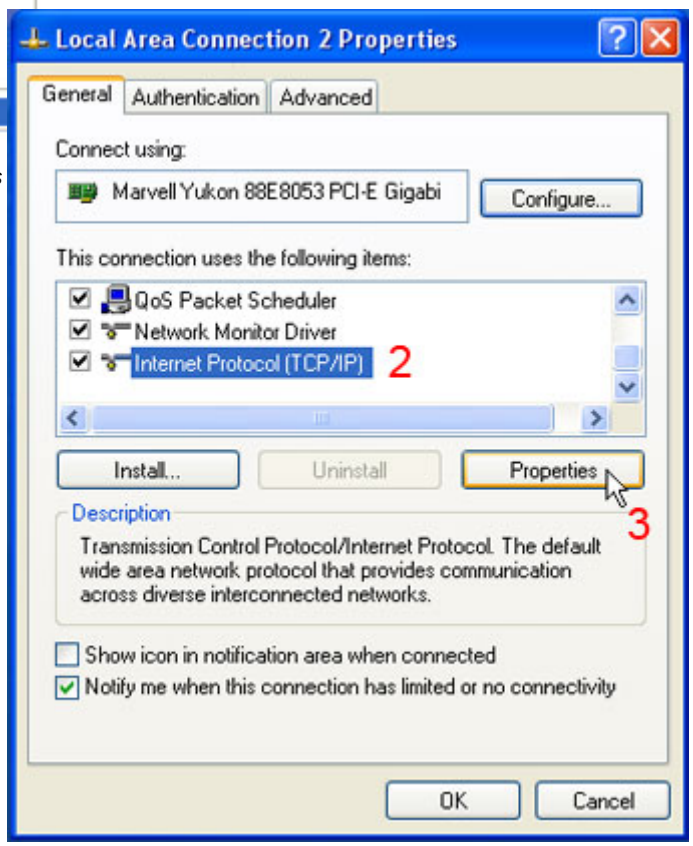

Fig. 6.3 Local Area Connection - Properties - TCP/IP Properties

How to set your PC's IP address continued :

4. On the TCP/IP Properties dialog :
If you wish to set your PC to obtain an IP address from the network's DHCP server :
Click the radio button shown.
Then Click the OK button (6)
Then click the Close button on the former Network Connections dialog. Then close the Local Area Connections window.
This should be done whilst connected to a LAN so the DHCP server is available and an address is granted. If not connected then there are other methods - but the easiest is to connect to the LAN and reboot.

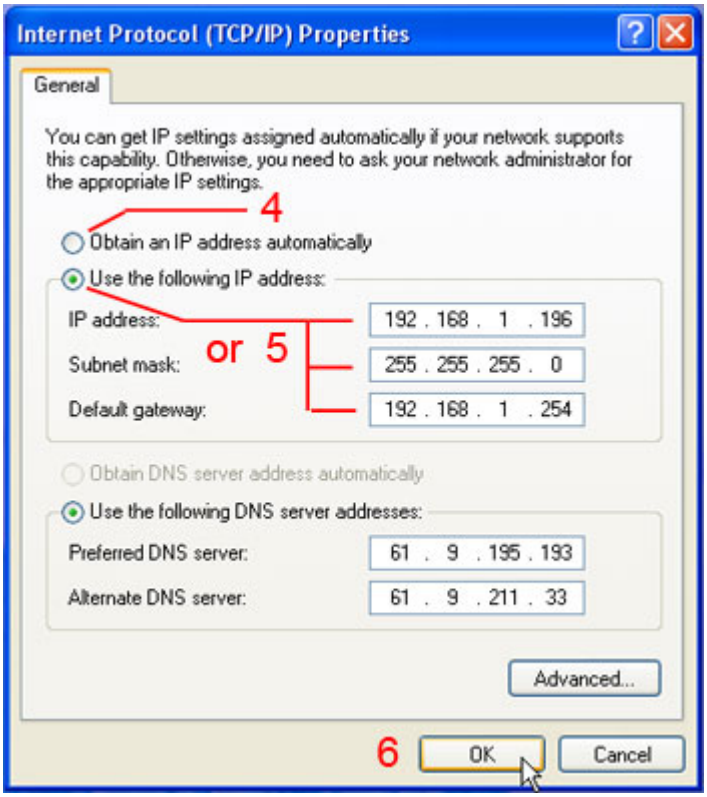

Fig. 6.4 TCP/IP Properties Settings Dialog (example only)

OR

5. If you wish to fix the IP of your PC
Click on the "Use the Following …" radio button. Then fill in the IP, Subnet Mask, and Gateway fields with the desired values.
then Click the OK button (6)
Then click the Close button on the former Network Connections dialog.
Then close the Local Area Connections window.
A short while later (after the hourglass cursor disappears) your PC will have the IP address you set.

How to Discover the IP address your PC has currently :

From the Desktop :
Click Start > Run > type "cmd"> click OK - the command console will be shown :
Type "ipconfig /all" and then tap the Enter key.
The PC's full IP configuration will be shown, including DNS servers etc.
This is a valid way to discover the network gateway if unknown and your PC is set to auto.

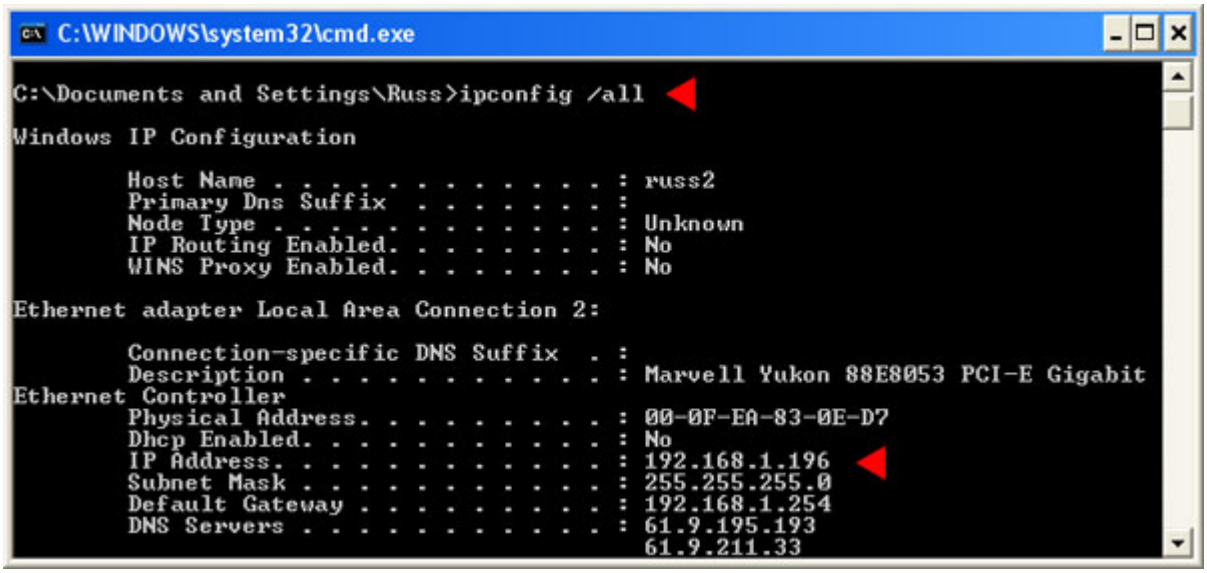

Fig. 6.5 MS Windows XP/2000 Command console (example only)

Appendix B:- ISU3 Serial Console Method : Using HyperTerminal

The Serial Console on the ISU3 (in fact all VIU platform cards) is a very useful way to achieve many settings without having to access the cards Web console. Items such as the cards IP address, subnet, gateway, initially; as well as many more buried features.

To use the serial console the first thing you will need , besides your PC, is a Hybrex Serial Cable - the same kind that you would use to upgrade a G1 or G2 for example.

the next step is to launch and configure a console application - Windows has such an application built in : it's called HyperTerminal - and you will find it, from the desktop, at : Start > Programs > Accessories > Communications > HyperTerminal.
It is a very good idea to right click this menu item and "Send to Desktop (create shortcut)" as you will find many uses for it.

When HyperTerminal is launched it will initially display a New Connection dialog :

Fill in a name for the configuration you are about to make that makes sense for you, and click OK
Later on after having saved this "connection" the next time you launch HyperTerminal you can cancel the New Connection dialog - and then "Open" your saved Config. - this saves time.

Fig. 6.6 HyperTerminal - New Connection Dialog

Fig. 6.7 HyperTerminal - Connect To Dialog

On the next dialog presented "Connect To" :
Select the Com port you will be using.
There will usually only be one.

As a point of interest : you will notice you can also select a TCP port - this has use if you wanted to observe the SMDR output of a GDS on its LAN port : port 1024 for example.

On the next dialog presented " Com x Properties" :
Set the Bits per second to 9600
The rest can be left at default but out of interest the values are 8, n, 1 and flow control as Hardware or None.

Click the OK button :-
The HyperTerminal window will be shown.

Connect your serial cable between the PC's COM port and the serial port on the ISU3 (there is only one RJ12 connector - which is the serial port)

When you are finished - later on - you may wish to "save" the connection when you close HyperTerminal for the reasons mentioned above.

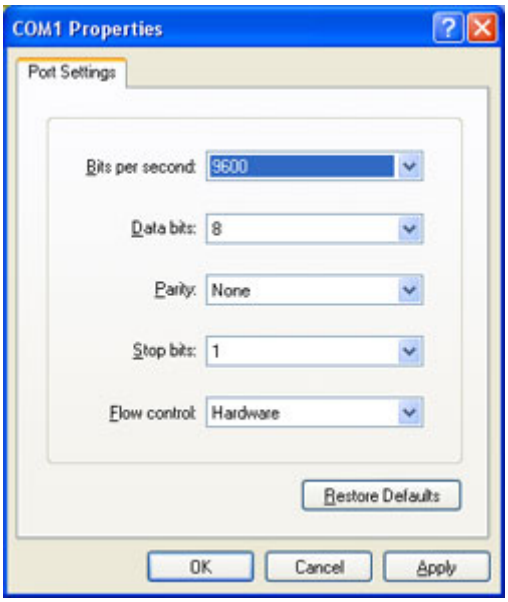

Fig. 6.8 HyperTerminal - Com1 Properties

ISU3 Serial Console : The ggdbg> prompt

With the ISU3 serial connection in place (and everything powered up of course) focus the HyperTerminal window (click in it) and tap the PC's Enter key - (noted following as [Enter]): The ISU3 's ggdbg> prompt will be shown:

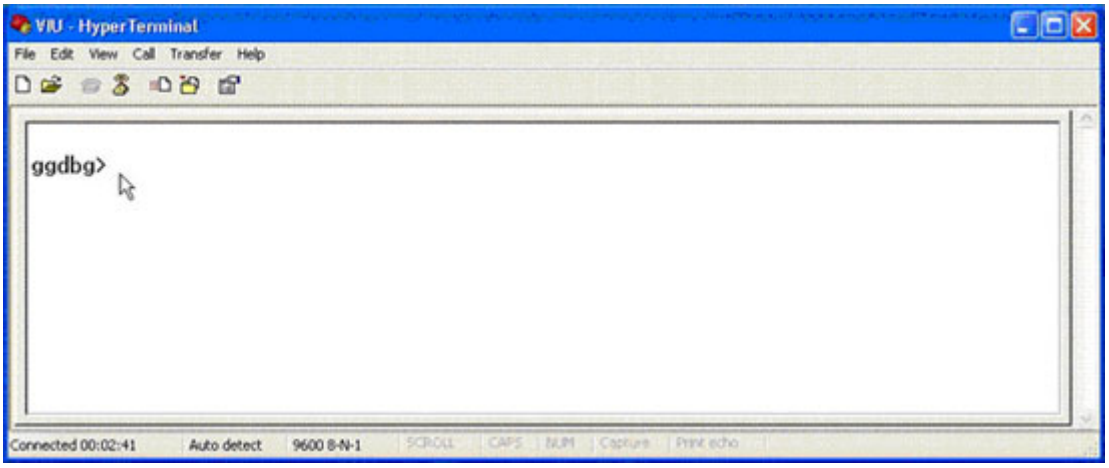

Fig. 6.9 HyperTerminal Console - ISU3 ggdbg> prompt

I call this the "Golden Gateway Debug Prompt" for various reasons but that's just an aside. This prompt is where you will issue commands by typing certain strings, then tapping your [Enter] key.

Discovering / Setting the ISU3 IP address

The first and most common thing that can be done is discovery and/or setting of the ISU3 IP address, and associated details :

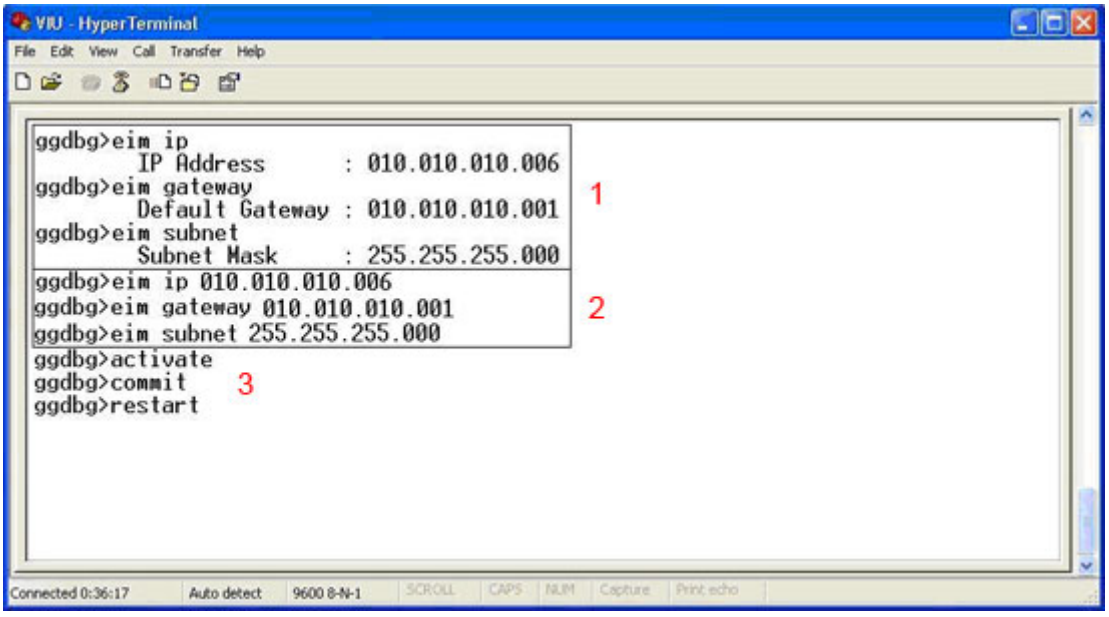

Fig. 6.10 HyperTerminal Console - ISU3 - eim commands

1. Typing the commands : "eim IP", or "eim gateway", or "eim subnet", followed by [Enter] will display the ISU's current IP details accordingly.

2. Set the IP address etc by following the above commands with the desired address etc For example typing "eim ip 10.10.10.6" [Enter] will set the IP address to that value.

3. To be applied properly any settings made above must be followed by the commands : "activate" [Enter], "commit" [Enter], "restart" [Enter], in that order.
The ISU will then restart with the new parameters in place.

Other Common ISU3 Serial Console Commands

Before detailing other common console commands for the ISU3 there is one thing that should be noted about method : The ISU3 console, as you will notice when connected, is fairly busy at times outputting the card status as various functions are enacted. This output stream is separate to any input you make - except for the responses to your input. When you are typing in commands you cannot be sure the display of the string you are entering will not be interrupted by an on-screen status output. The effect of any such interruption is that you may have the impression that your input is lost - it is not : as noted the in and out streams are separate. So when entering a command you primarily need to ignore any status outputs and just keep typing your command string. If you get a command spelling wrong the card will tell you with "Unrecognised" or other so you should look at the screen immediately after you send the command with your Enter key. Otherwise if you are really unsure about the entry of an IP or other number then you can enter the command and details again.

Factory Default

If for any reason - eg. perceived corruption - you wish to factory default a VIU / ISU3 card issue the command :

```
ggdbg> set default factory [Enter]
```

The card will then reboot loading the factory defaults in the process.
Note all settings you have made will be lost so these need to have been noted beforehand.
The factory default IP of the card is 10.10.10.6.
You can then set the desired IP from the serial console (see previous) and proceed from there.

Recovery from an Inadvertent PPPoE Setting

If you have inadvertently set the ISU3 to PPPoE mode, or when in PPPoE mode the ISU3 cannot log on because of a password mistake or other, the ISU3 Web console will not be available because the card will end up with an IP of 0.0.0.0.
In this case a recovery method is to use the serial console to reset the "dhcp" mode (PPPoE is just another form of DHCP) : Issue the command :

```
ggdbg> set dhcp off [Enter]
```

This will leave all other details except IP settings intact and so needs to be followed by resetting the IP to a value that allows you to access the Web console again. viz:

```
ggdbg> eim ip xxx.xxx.xxx.xxx [Enter]
ggdbg> eim subnet xxx.xxx.xxx.xxx [Enter]
ggdbg> eim gateway xxx.xxx.xxx.xxx [Enter]
ggdbg> activate [Enter]
ggdbg> commit [Enter]
ggdbg> restart [Enter]
```

If you want PPPoE mode and think, or know, that the error stopping the card attaining a login is a mistake in either the username or password you have entered then before reverting the card as above you could try the following :
To correct the PPPoE username :

```
ggdbg> set pppoe_username {enter correct username} [Enter]
```

To correct the PPPoE password :

```
ggdbg> set pppoe_password {enter correct password} [Enter]
```

Follow these commands with the **activate, commit, restart** sequence as above.

Setting Silence Detection

Silence detection for VoIP is a process where the RTP bandwidth required for a VoIP call is reduced by the device not sending a full voice stream for periods where the senders audio activity is below a certain threshold (periods of silence). This is possibly useful to some users where bandwidth is at a premium, but the undesirable effect is the receiver hears nothing for these detected silence periods and the perception is often that the line has gone "dead". Another noticed effect is that when silence detection is active it takes a moment to

restart the voice stream so there is often the effect of “clipping” of the first word spoken after a silence. Both these effects have drawn complaints from customers.
Up to the current firmware version of the ISU3 (VIUSXO-009) silence detection is enabled for all codec profiles available except G729. This silence detection can be disabled profile by profile, as desired, via the serial console. The commands to use are shown below.
The ISU3 codec profile numeric order is : profile 1 = G723 (6.3K), profile 2 = G729ab, profile 3 = G711u, profile 4 = G711a, profile 5 = T.38 fax.
To control silence detection for any particular codec issue the following command:

```
ggdbg> set coding {profile No.} vad {on/off} [Enter]
```

Where off = disable etc. When you have set the codec profiles follow with :

```
ggdbg> activate [Enter]
ggdbg> commit [Enter]
ggdbg> restart [Enter]
```

Using the Serial Console During Firmware Upgrades

A point not covered in the ISU3 section is the firmware upgrade process. Ideally this process is fairly simple using the Web Console and MS Internet Explorer browser.
At certain times however, for example when access to the Web console is lost, the serial console can be used to force feed firmware into the card. The details of both methods are not covered here, but are available in a document called “VIU Upgrade Procedure Notes” which is available from Hybrex Support and/or the Aust. Hybrex dealer FTP site.

Appendix C:- Voice Codecs for ISU3

To enable voice to be transmitted as a digital data stream the analogue form is first digitized by an A to D converter. This data stream is then passed to a codec (short for coder-decoder) to compress the voice data to reduce the bandwidth needs of the stream. The compressed voice data is then assembled into packets for transmission to the destination using the Internet Protocol stack. The packetising process introduces an overhead in terms of bandwidth required as the voice data is "wrapped" in a lot of other data (source and destination IP addresses, packet sequence count and size, DSCP markers, CRC codes, etc. etc.) that is part of the Internet Protocol and enables the transport method. The other point worth a mention here is that the SIP protocol used to form the voice connections, and the RTP protocol used to transport the voice data, both use the UDP (Universal Datagram Protocol) as their transport method. UDP is "connectionless" in that the packets are just sent - there is no error checking or re-send requests as there is for TCP (TCP is a connection based protocol), which makes sense when you consider that resending a voice packet is redundant due to the "real time" nature of the stream. There are exceptions appearing recently however where for example SIP connections to Microsoft Exchange servers are formed over TCP.

The ISU3 has 5 codec profiles available - as mentioned previously - so these will be defined. The codecs available are : G723 (6.3K), G711u, G711a, G729ab, and T.38.(untested) When setting the codec on the ISU3 DSP Settings page you are only indicating the preferred codec (first in the preferences list) for use in any voice connection. The other codecs available may be used instead as will be determined by the capabilities of the receiving device and determined during the SDP negotiations that occur at call setup.

Bandwidth Requirements

The bandwidth requirements for each voice stream according to codec in use can be broadly stated as:

Codec	Raw data	Packetised (IP) data	Notes
G711a	64Kbps	90Kbps	Considered "Toll Quality"
G711u	64Kbps	90Kbps	Considered "Toll Quality"
G729a/b	8Kbps	30Kbps	Best BW/Quality tradeoff, most popular
G723 (6.3)	6.3Kbps	22Kbps	OK for LAN work - use if desperate

When considering actual bandwidth requirements, the figures above must be multiplied by the number of concurrent voice connections at any time, and a margin needs to be added for SIP and other network traffic (say 5~10%).
A note : anecdotes are that G729 is much better at coping with network packet loss (re voice quality) than G711.

Codec acceptance with regard to network issues (one way packet delay) can be summarized by the graphic :

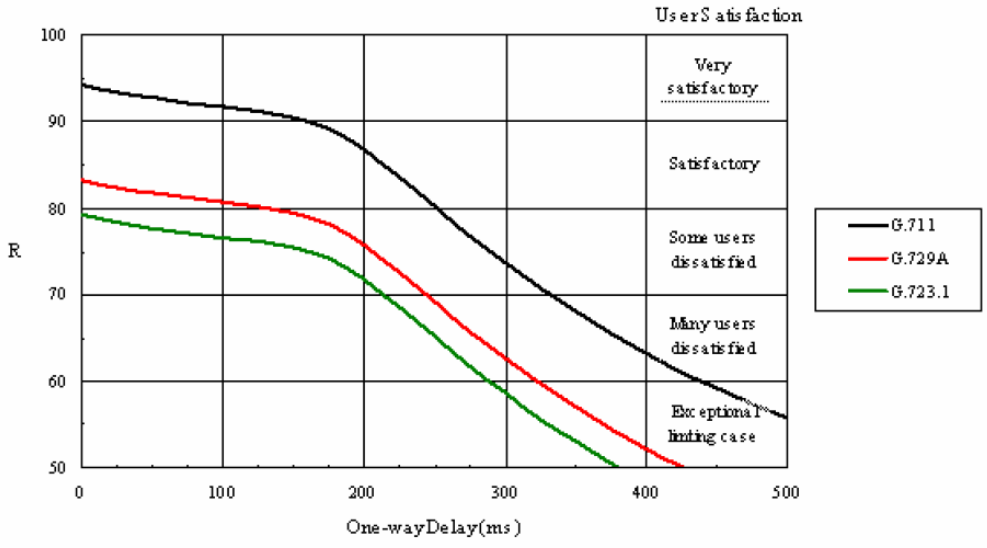

Fig. 6.11 Voice Codec Comparison - Re packet delay

Voice Frame Sizes

Another factor relative to codec setups is the voice frame size ie.: how much voice is sent in each packet. Whilst the SIP protocol allows anything from 10mS to 160mS of voice sent per packet industry norms are a little more conservative.
The standard is generally accepted at 20mS voice per packet.
Other packet sizes in use are 10mS, 30mS, 40mS, and even in some cases 60mS.
Refering to the ISU3 DSP Settings Page the settings there will provide :
Codec set = G711 VFP Small = 10mS, VFP Med = 20mS, VFP Large = 30mS
Codec set = G729 VFP Small = 20mS, VFP Med = 30mS, VFP Large = 60mS
The tradeoff when setting Voice Frame sizes is : the larger the voice frame the fewer and larger the packets so there may be a network competitive edge - But : the faster voice quality is affected in the negative if packets are being dropped or don't arrive in time for inclusion in the receive buffer.

Receive Buffer Sizes

All devices that handle RTP voice streams have a receive buffer where the RTP voice packets are assembled on reception before being forwarded to the receiving codec and subsequent voice reproduction hardware. This is in effect a jitter buffer as RTP voice packets will arrive at differing time intervals according to network traffic, and may even arrive out of sequence if different network paths are taken. So setting a larger receive buffer is better insurance against untimely packet arrival and consequent voice quality drop when packets don't arrive in time for inclusion in the processed voice stream. The tradeoff is a larger receive buffer contributes directly to packet delay - see previous time vs quality table.

Data Requirements Over Time for Voice

As mentioned in the routers section: Data requirements over time will need to be estimated when initially considering broadband supply. Beyond initial estimates monitoring and account adjustment should be considered. Data needs can be initially estimated using a formula based on anticipated talk time and codec choice. Say you use the G729 codec (30kbps/sec/channel) and anticipated voice use is 2 Hrs/day on each of the three ISU3 channels, data requirements would look something like : ((((30k/8) x 60 x 60) x 2)x 3) = 81 MBytes per day voice plus a notional 20% network overhead = 97.2MB/day or ~~2.2 GB per month (22 wkg days).

Appendix D:- Dial Plans

Many SIP devices these days have various gateway use controls that are based on the device requesting the connection or the number dialed. In the case of the number dialed the controlling rule set is commonly known as a "Dial Plan". There is a fairly standard syntax for dial plans in use today. One of the devices described in this guide is the SPA3102 which uses dial plans, and as a matter of interest more information regarding Dial Plan syntax is included here FYI. This information is lifted directly from the short form manual for the SPA3000 series written by Jason of JMG Technology so full credit goes to him for the following :

Dial plans can be very confusing at first glance. However they are invaluable feature of the SPA-3000 so you should at least learn the basics of how they work. The following dial plans are to show how to use the various features of dial plans. You should play around with them to suit your needs.

Dial Plan Syntax

( ) - The entire dial plan must be surrounded by an open and close bracket.
| - Each individual dial plan must be separated by a pipe | character.
0-9 - Treated as normal digits
x - Treated as any normal digit 0-9 on phone
***** - Treated as normal * character on phone
- Treated as normal # character on phone
. - Repetition
< : > - Replacement, eg <02:612> means replace 02 with 612
<:@gw0> - Gateway 0 is the PSTN line
<:@gw1> - Gateway 1 (Advanced Feature)
<:@gw2> - Gateway 2 (Advanced Feature)
<:@gw3> - Gateway 3 (Advanced Feature)
<:@gw4> - Gateway 4 (Advanced Feature)
S0 - Dial Immediately
! - Barring a number, place this at the end of the number to bar it
, - Provides a dial tone
[ ] - Limiting choices, eg [24] means either 2 or 4, [2-5] means 2,3,4 and 5, [24-68] means 2,4,5,6,8

Example Dial Plans

Dial Plan 1: (000S0<:@gw0>)
Description: The above dial plan is extremely simple, yet extremely important. When you dial 000 (Emergency number) your call will go out through Gateway 0 (<:@gw0) which is your normal PSTN line, immediately (S0) after you have dialed the 3rd 0.

Dial Plan 2: (000S0<:@gw0>|1800xxxxxxS0<:@gw0>)
Description: The above dial plan contains two individual plans, building on from DialPlan 1. You will notice that a | separates the 1st dial plan from the 2nd .The 2nd dial plan is used to route 1800 numbers through the your PSTN line. It works the same way as the 1st dial plan, in that when you dial a 1800 number followed by 6 other digits (0-9) it will be directed through your PSTN line.

Dial Plan 3: (<*1:0123456789>)
Description: This plan demonstrates replacement. If you dial a * followed by a 1 then the number 0123456789 would be dialed.

Dial Plan 4: (<0:61>[2-9]xxxxxxxxS0)
Description: This plan demonstrates replacement and limiting choices. When you dial an 0 followed by a 2,3,4,5,6,7,8 or 9 and then nine of any other digit (0-9) it will prepend 61 and remove the 0. So if you rang 02 123456789 the actual number that would be called would be 61 2 123456789.

Dial Plan 5: (1900xxxxxx!)

Description: This plan demonstrates number barring.
If you enter a 1900 followed by 6 more digits (0-9) you call will not be placed.

Dial Plan 6: (<#9:>xx.<:@gw0>)

Description: This plan demonstrates replacement and repetition.
When you enter a #9 followed by any number of digits (a timeout is used to determine the end) it will go out through the PSTN line (Gateway 0).

Putting it all together

Dial Plan 7: (000S0<:@gw0>|1[38]xxx.<:@gw0>|1900xxxxxx!| 0[2-9]xxxxxxxS0| <#9:>xx.<:@gw0>)

Description: This plan combines elements from all the above dial plans. It routes all 000, 1800, 1300 calls out via the PSTN line. Its bars 1900 numbers. It allows an Australian land line to be called and it also allows you to select the PSTN line by dialing a #9.

Appendix E :- Dynamic DNS

Just a short not for now on Dynamic Domain Name Services. Or DDNS for short.

Many Internet supply accounts are “dynamic” in that every time the connection (ISP login) is broken and reformed - by a router reset for example - the account is given a different WAN IP. This makes things a tad difficult for running any kind of server for example that is accessible from the Internet as the IP to access it keeps changing.

Dynamic DNS is a service whereby a DDNS enabled router will report any change in its WAN IP address to an external entity - a DDNS server - so that the name registered on that server for the “account” that the router reports with is updated with the new WAN IP.

The formation of DDNS services means a broadband Internet account that only has a dynamic WAN IP allocation can enjoy the luxury of a constant fully qualified domain name (FQDN) that links to their WAN port whatever its IP address is.

The Draytek routers described in this guide are DDNS enabled.
You will find their “DynDNS” setup page under the Applications menu link.
Log onto the Draytek Web console and go to the DynDNS page and note the second level domain name options available - you will need to do this before getting a DDNS account as these name choices are fixed in the router so you need to get an account name you can actually set up in the router.

There are a number of Dynamic DNS service providers in the Internet.
Some of them offer free services :
One such provider can be found at http://www.dyndns.com
If you want to use this service :
Browse to dyndns.com and then click the “Account” link on the central menu bar.
On the next page provided click on “Create an Account”
Then fill in the requested details, login username and password, specify your preferred domain name, select a second level domain name from the choices provided (to match one of the options from the Draytek above), and complete the account registration.
The domain name you get to use will follow a format like [yournamechoice].dyndns.org

You will notice dyndns.com allows you to register 3 domain names for free - the caveat is that you must “renew” any domain name (refresh the associated WAN IP) every 30 days or the account will be “flushed” - general housekeeping that makes sense re server resources. They do offer email notification of impending action however so are very fair about it.

You then fill in the relevant details on the Draytek DynDNS application page, enable it, force an update, and you should be “live”.

Although not recommended due to added complexity and inherent connection formation times the DDNS process above has been tested with an ISU3 and found to be operational. In short the ISU3 was situated behind a Draytek router (2700e) on a dynamic IP and the ISU3 Outbound Proxy and Registrar IP field values set to the DDNS FQDN. SIP Phones were also set up with accounts that stated the DDNS FQDN as Registrar and Proxy addresses. Registrations were immediate and calls made and accepted without issue.
In all cases DNS functions were enabled and the local DNS server addresses specified.